L Number	Hits	Search Text	DB	Time stamp
1	167	auxiliary adj0 service	USPAT	2004/03/11 18:23
2	0	"out of band"	USPAT	2004/03/11 16:12
3	43942	379/$.ccls.	USPAT	2004/03/11 16:13
4	20	(auxiliary adj0 service) and 379/$.ccls.	USPAT	2004/03/11 16:16
5	48733	audible	USPAT	2004/03/11 16:16
6	2444	"in band"	USPAT	2004/03/11 16:16
7	51035	audible "in band"	USPAT	2004/03/11 16:16
8	0	(auxiliary adj0 service) same (audible "in band")	USPAT	2004/03/11 16:16
9	15	(auxiliary adj0 service) same channel	USPAT	2004/03/11 18:22
10	2	379/$.ccls. and ((auxiliary adj0 service) same channel)	USPAT	2004/03/11 16:22
11	5	prohs.in.	USPAT	2004/03/11 16:22
12	22	transparent$2 with "in band"	USPAT	2004/03/11 16:22
13	3	379/$.ccls. and (transparent$2 with "in band")	USPAT	2004/03/11 16:23
14	6	transparent$2 with inband	USPAT	2004/03/11 16:41
15	0	(auxiliary adj0 service) same inband	USPAT	2004/03/11 16:25
16	1751	inaudible	USPAT	2004/03/11 16:25
17	0	(auxiliary adj0 service) same inaudible	USPAT	2004/03/11 16:25
18	3166	inband or "in band"	USPAT	2004/03/11 16:41
19	285841	transparent$2	USPAT	2004/03/11 16:41
20	16	(inband or "in band") near2 transparent$2	USPAT	2004/03/11 16:43
21	0	transparent$2 near2 siganl$4	USPAT	2004/03/11 16:43
22	2449	transparent$2 near2 signal$4	USPAT	2004/03/11 16:44
23	3	(auxiliary adj0 service) and (transparent$2 near2 signal$4)	USPAT	2004/03/11 16:46
24	115	379/$.ccls. and (transparent$2 near2 signal$4)	USPAT	2004/03/11 16:47
26	1	(transparent$2 near2 signal$4) and 5267305.uref.	USPAT	2004/03/11 16:47
27	4	(transparent$2 near2 signal$4) near2 dtmf	USPAT	2004/03/11 16:47
25	12	5267305.uref.	USPAT	2004/03/11 17:02
29	0	5267305.pn. and isdn	USPAT	2004/03/11 17:03
30	1	5267305.pn. and channel	USPAT	2004/03/11 17:34
31	18	(transparent$2 near2 signal$4) same isdn	USPAT	2004/03/11 17:34
28	1	5267305.pn.	USPAT	2004/03/11 18:02
32	519	auxiliary adj0 service	USPAT; US-PGPUB; EPO; JPO; DERWENT; IBM_TDB	2004/03/11 18:23
33	1	transparent$2 same (auxiliary adj0 service)	USPAT; US-PGPUB; EPO; JPO; DERWENT; IBM_TDB	2004/03/11 18:27
34	5028	inband "in band"	USPAT; US-PGPUB; EPO; JPO; DERWENT; IBM_TDB	2004/03/11 18:28
35	16	transparent$2 near2 (inband "in band")	USPAT; US-PGPUB; EPO; JPO; DERWENT; IBM_TDB	2004/03/11 18:29
36	0	(auxiliary adj0 service) same (inband "in band")	USPAT; US-PGPUB; EPO; JPO; DERWENT; IBM_TDB	2004/03/11 18:29

US005937040A

United States Patent [19]

Wrede et al.

[11] **Patent Number: 5,937,040**

[45] **Date of Patent: Aug. 10, 1999**

[54] **METHOD AND APPARATUS FOR USING A D-CHANNEL FOR DISPLAYING USER DATA**

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[73] Assignee: **Siemens Information and Communication Networks, Inc.**, Boca Raton, Fla.

[21] Appl. No.: **08/844,417**

[22] Filed: **Apr. 18, 1997**

[51] **Int. Cl.**[6] **H04M 11/00**

[52] **U.S. Cl.** **379/93.23**; 379/93.17; 379/100.01; 370/524

[58] **Field of Search** 379/93.01, 93.05, 379/93.09, 93.17, 93.23, 93.14, 93.15, 100.01, 100.13; 370/522, 524, 465, 466, 467

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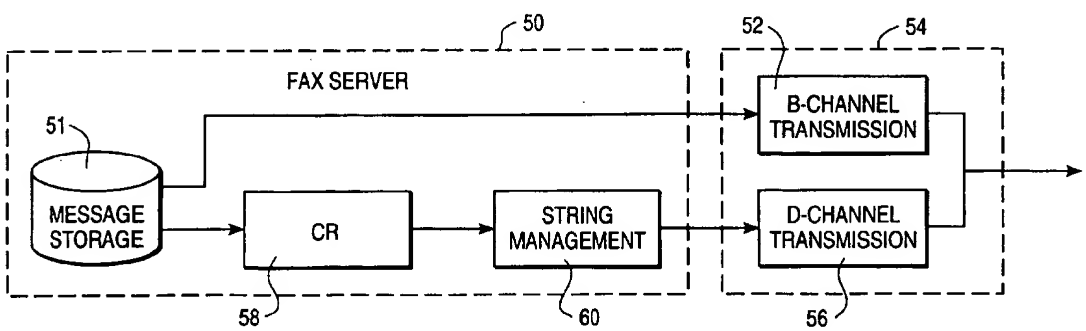
Primary Examiner—Curtis A. Kuntz
Assistant Examiner—Melur Ramakrishnaiah

[57] **ABSTRACT**

A telecommunications method and apparatus for sending non-protocol messages from a source to a remote phone terminal having a display includes converting the messages at the source into a format that is compatible with transmission via a signaling channel of a telecommunications line having at least one digital user data channel in addition to the signaling channel. For example, the telecommunications line may be an ISDN link. The formatted messages are visually displayed at the remote phone terminal. As examples, the messages may be menu options of an interactive voice response unit, facsimile documents of a facsimile server, or voicemail messages of a voicemail server.

20 Claims, 5 Drawing Sheets



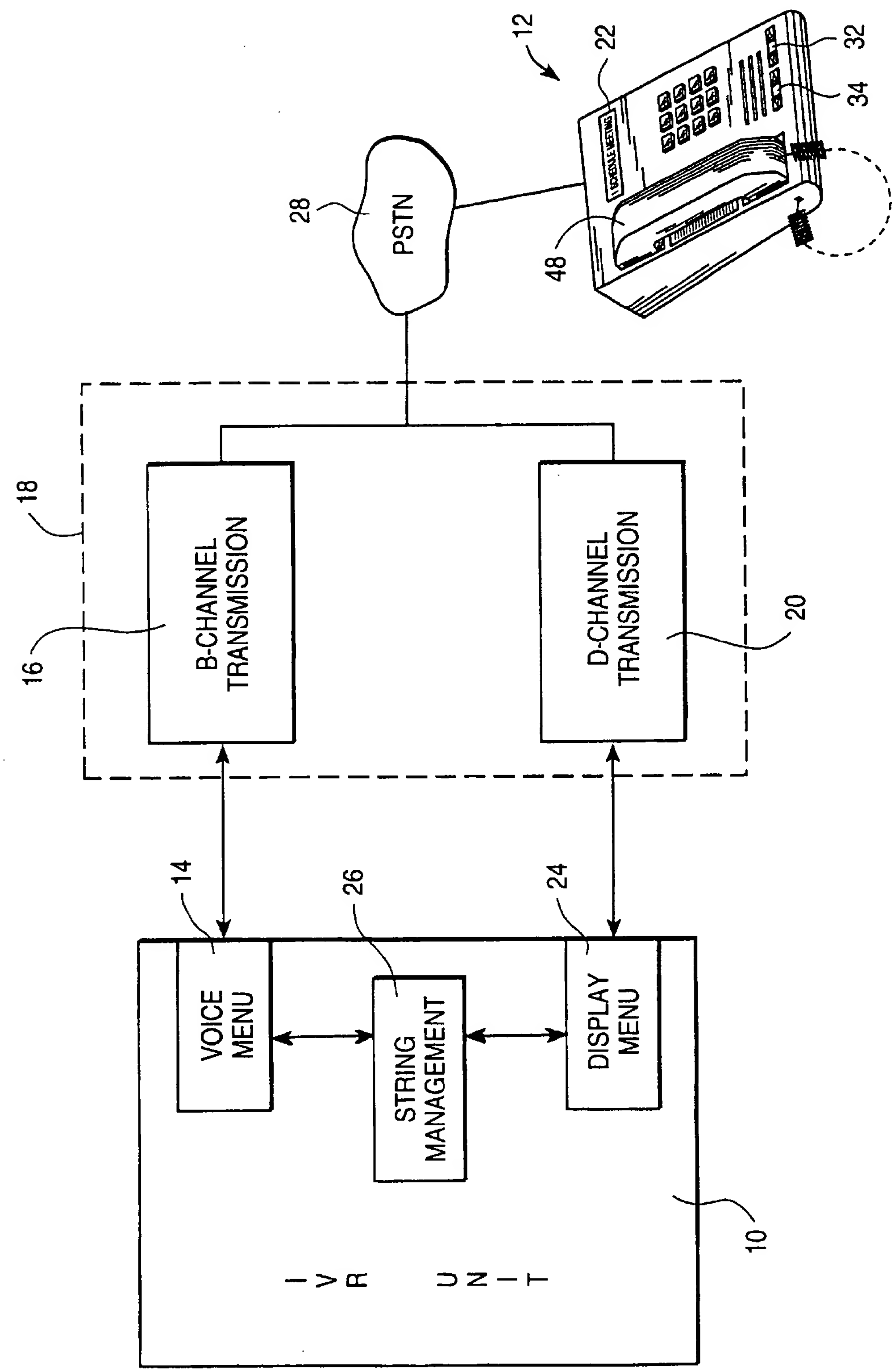

12
22
32
34
28
PSTN
48
18
B-CHANNEL TRANSMISSION
16
D-CHANNEL TRANSMISSION
20
14
26
24
VOICE MENU
STRING MANAGEMENT
DISPLAY MENU
IVR UNIT
10

FIG. 1

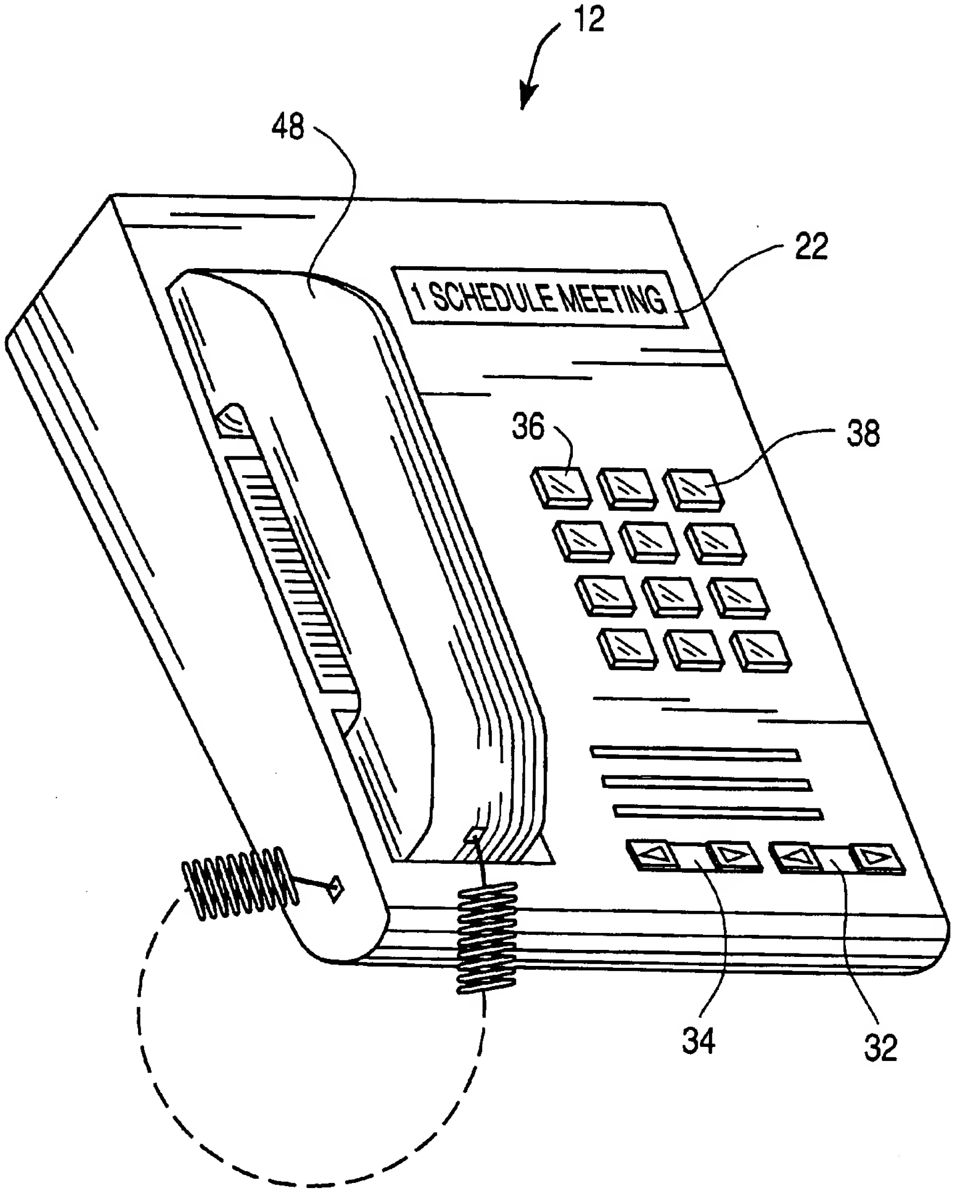


FIG. 2
(PRIOR ART)

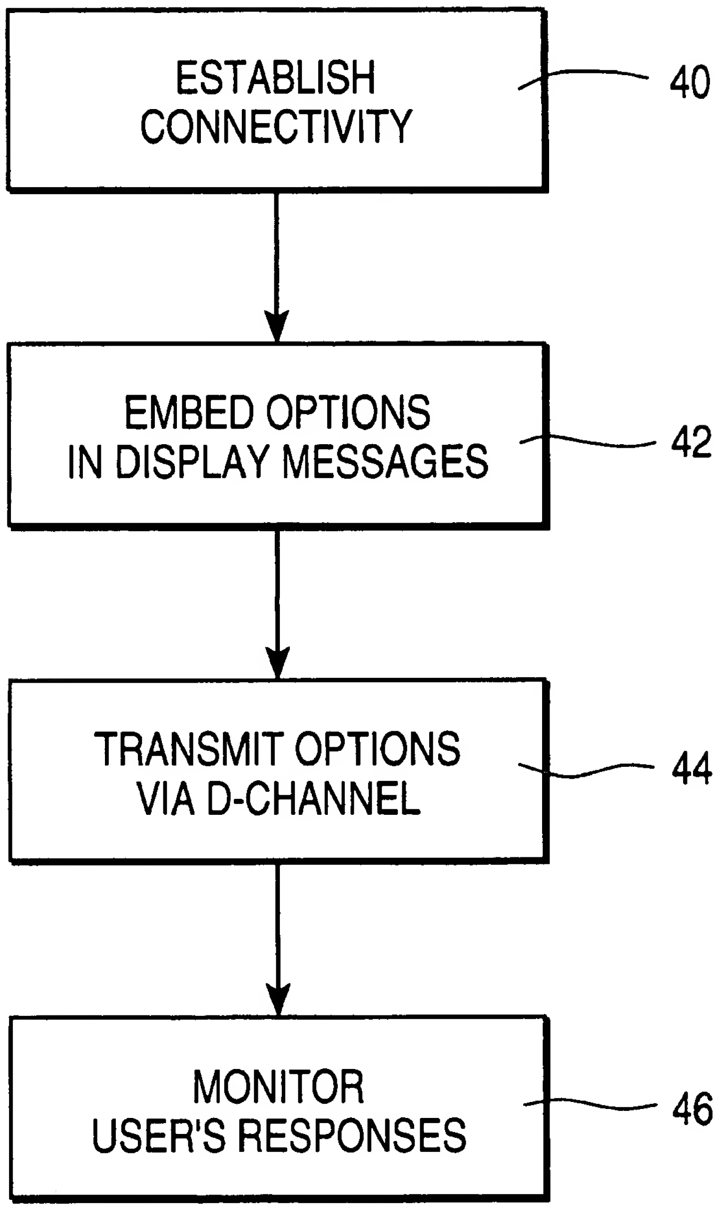

ESTABLISH CONNECTIVITY
40
EMBED OPTIONS IN DISPLAY MESSAGES
42
TRANSMIT OPTIONS VIA D-CHANNEL
44
MONITOR USER'S RESPONSES
46

FIG. 3

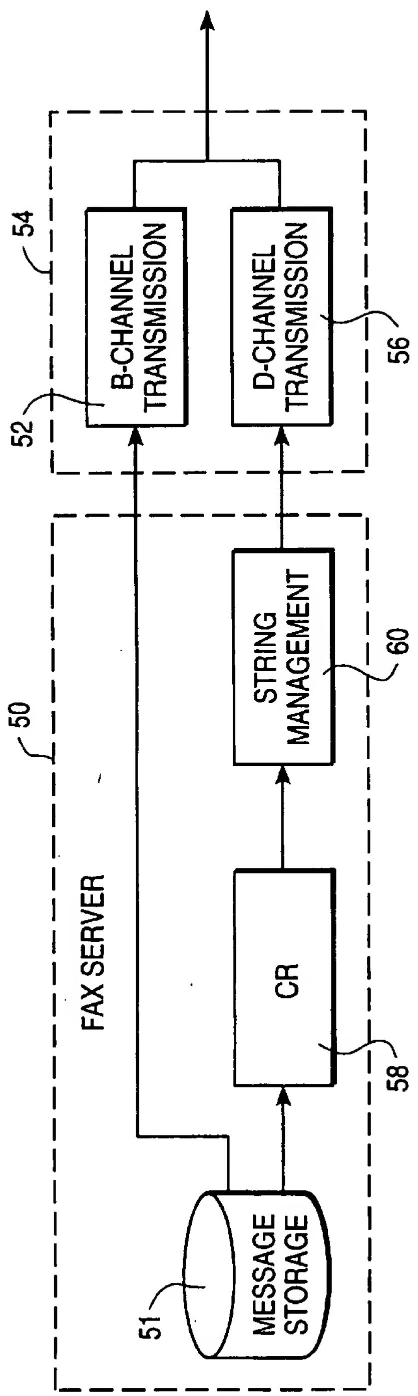

FAX SERVER
MESSAGE STORAGE
CR
STRING MANAGEMENT
B-CHANNEL TRANSMISSION
D-CHANNEL TRANSMISSION
50
51
52
54
56
58
60

FIG. 4

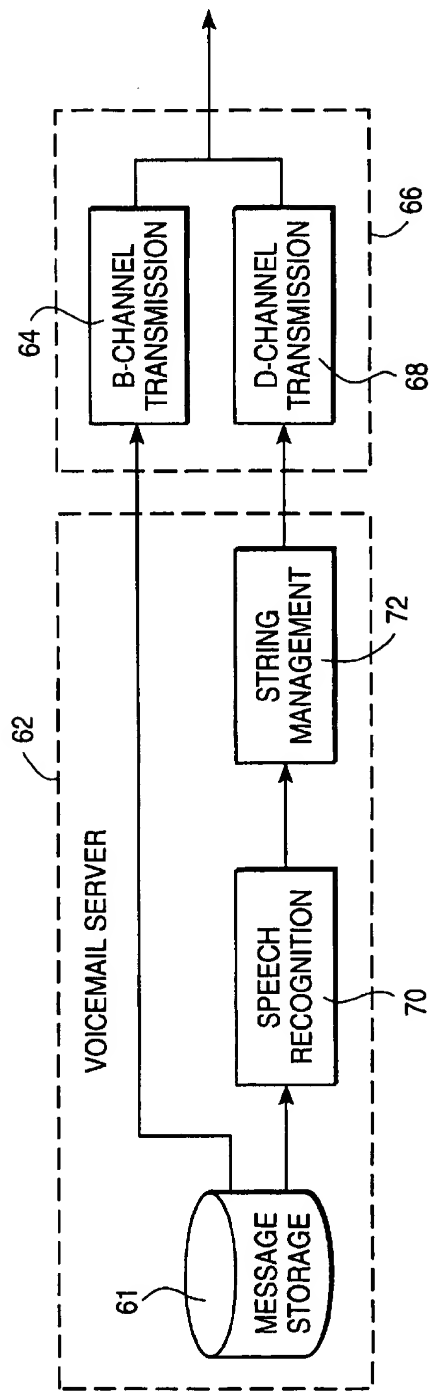

VOICEMAIL SERVER
MESSAGE STORAGE
SPEECH RECOGNITION
STRING MANAGEMENT
B-CHANNEL TRANSMISSION
D-CHANNEL TRANSMISSION
61
62
64
66
68
70
72

FIG. 5

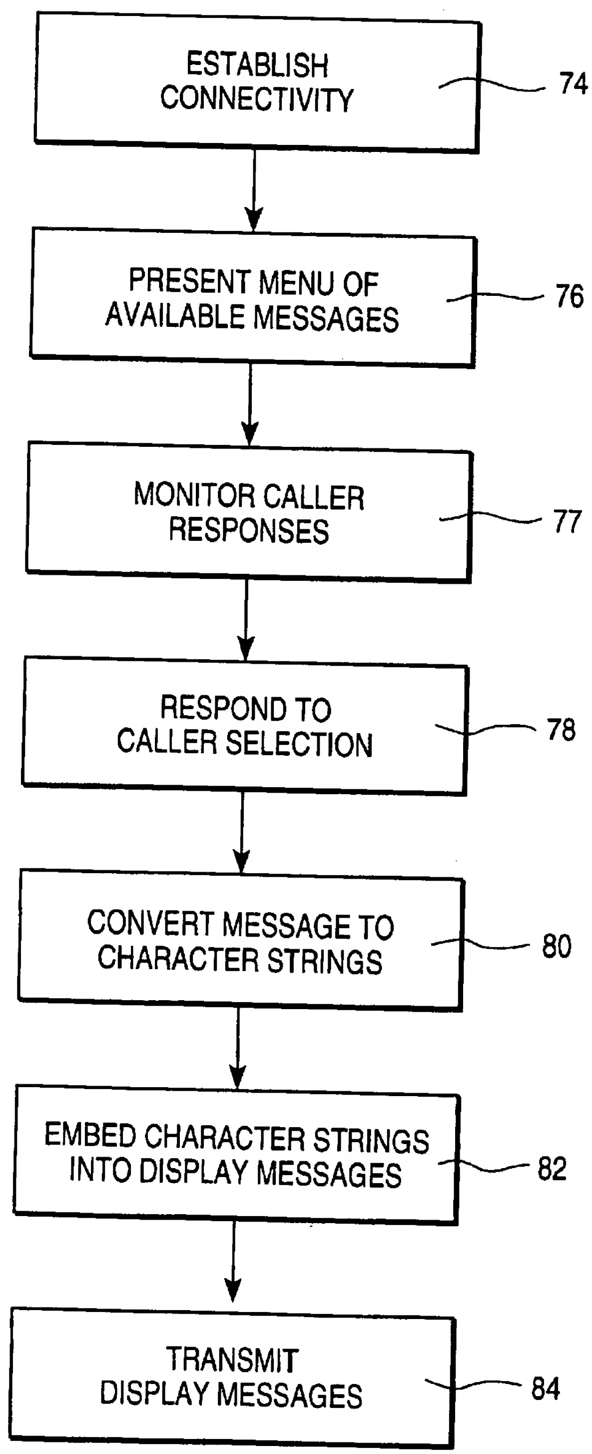

ESTABLISH CONNECTIVITY
74
PRESENT MENU OF AVAILABLE MESSAGES
76
MONITOR CALLER RESPONSES
77
RESPOND TO CALLER SELECTION
78
CONVERT MESSAGE TO CHARACTER STRINGS
80
EMBED CHARACTER STRINGS INTO DISPLAY MESSAGES
82
TRANSMIT DISPLAY MESSAGES
84

FIG. 6

METHOD AND APPARATUS FOR USING A D-CHANNEL FOR DISPLAYING USER DATA

BACKGROUND OF THE INVENTION

The invention relates generally to methods and apparatus for presenting messages to a party via a telephone network and more particularly to transmitting messages using a digital communications line having at least one user data channel and a separate signaling channel.

DESCRIPTION OF THE RELATED ART

Digital telephones provide features that are typically not available to a user of a conventional analog telephone. The higher bandwidth that is available when using digital communications lines accommodates the addition of advanced features. An Integrated Services Digital Network (ISDN) includes at least two B-channels that carry user data and includes a D-channel that carries signaling information. Each of the B-channels is a 64 Kbps digital channel, while the D-channel may have a bandwidth of 16 Kbps or 64 Kbps. A Basic Rate Interface (BRI) has two B-channels and a single Kbps D-channel, and is therefore sometimes referred to as a 2B+D interface to ISDN. The effective bandwidth of a BRI is 144 Kbps. Even greater bandwidth can be achieved using a Primary Rate Interface (PRI) that consists of either 23 or 30 B-channels for user data and a single 64 Kbps D-channel for the signaling information. The 23B+D interface is used in North America and in Japan to provide an effective bandwidth of 1.544 Mbps, while the 30B+D method is used in Europe to provide an effective bandwidth of 2.048 Mbps.

Most of the signaling information that is passed along a D-channel is connectivity-related. That is, the D-channel carries call messages that are often transparent to the parties and relate directly to telephony signal exchanges. Examples of connectivity-related messages that are transmitted via the D-channel include CONNECT, CONNECT ACKNOWLEDGE, DISCONNECT, and RELEASE signals. However, not all of the connectivity-related messages are transparent to the parties of a call. Some digital telephones include display capability. Information regarding an incoming call may be displayed on a readout, such as a Liquid Crystal Display (LCD). This displayed connectivity-related information may include an identification of the telephone number and even the name of the calling party. Moreover, if an incoming call has been sent, the connectivity-related message may include the originally called telephone number and the reason for sending, e.g., sending from a ring-busy telephone or a ring-no-answer telephone.

The B-channels of an ISDN carry the user data. For ordinary telephone calls, the user data is digitized voice information. Digital transmissions are less susceptible to transport-induced distortion than analog transmissions. Additionally, any distortion that is induced is more easily filtered.

The user data that are transmitted via the B-channels of an ISDN may be facsimile data. Documents may be sent to a server and stored for later retrieval. An individual who identifies a particular document and/or provides a password is sent one or more documents. Conventionally, retrieval requires a facsimile machine or a personal computer having modem capability. Thus, the availability of the required equipment limits the ability of a user to download facsimile documents.

In addition to realtime voice information, user data transmitted via B-channels of an ISDN include recorded voice

information. For example, the B-channel voice information may be voicemail sent from a server. A voicemail server operates in a manner equivalent to the facsimile server. As another example of sending recorded voice information, voice prompts of an interactive voice response (IVR) unit are transmitted via B-channels. The IVR presents messages to a caller, with the caller being prompted to depress certain buttons on a keypad of a phone in order to make selections. For example, a pay-per-view facility may present a decision tree in which one menu level is used to specify a type of movie and a subsequent menu level is used to select a particular movie. A concern in the presentation of voice information to a caller is that it sometimes requires a relatively high degree of comprehension. Another concern is that a caller may sometimes have difficulty in hearing the voice information. For example, a caller in a noisy environment may miss portions of a voice message. In another example, the caller may be hearing impaired and, therefore, unable to use a pay-per-view facility or other IVR unit unless specialized equipment is accessed, such as a Telecommunications Device for the Deaf (TDD). However, such specialized equipment is often cost prohibitive.

What is needed is a method and apparatus for sending user data to increase accessibility to messages and to enhance reliability and comprehensibility of the messages.

SUMMARY OF THE INVENTION

A telecommunications method of sending user data during a call between a source of messages and a remote phone terminal having a display includes converting the messages at the source into a format compatible with transmission via a signaling channel of the telecommunications line having at least one digital user data channel in addition to the signaling channel. In an ISDN environment, the signaling channel is a D-channel and the user data channel is a B-channel. The user data sent via the D-channel define a message that is unrelated to forming, maintaining, or terminating the call. Nevertheless, the converted messages are transmitted via the signaling channel for presentation at the display, rather than the user data channel or channels.

In one embodiment, the converted user data are menu options of an interactive voice response unit. Optionally, a message is stored as both a text string in a text file and as voice information in a speech file that is associated with the text file. Within this multimedia embodiment, the text file is transmitted via the D-channel and the speech file is transmitted via the B-channel such that the visual presentation of the messages by means of the phone display is synchronized with the conventional audio presentation at the remote telephone. The conversion of the messages formats the information for the visual presentation at the display of the remote phone terminal. Again referring to the ISDN environment, the formatting may be achieved by embedding the menu information into DISPLAY Information Elements (IEs) or into containers (i.e., envelopes) of user-to-user information compatible with D-channel transmission, so that digital character strings are sent for display in realtime at a remote ISDN display phone. The visual presentation may be simultaneous with audio presentation of the menu options in order to increase comprehensibility or to accommodate the hearing impaired.

In another embodiment, the user data are facsimile documents and the source is a facsimile server. The conversion of the facsimile documents into a format for transmission via the signaling channel includes utilizing character recognition techniques. The reformatted user data can then be

embedded into the D-channel DISPLAY IEs or user-to-user IEs for realtime display on an ISDN display phone. Preferably, the facsimile server is responsive to signaling from the remote phone, so that the user determines the rate of transmission. The user may then scroll through the facsimile document.

In another embodiment, the user data are voicemail messages from a voicemail server. The conversion of the messages includes utilizing speech recognition techniques. For each of the facsimile and voicemail embodiments, a menu of the available messages may be displayed to the user prior to transmission, allowing the user to select among the available messages. For example, header information that identifies the content or the originating party of a facsimile or voicemail message may be embedded into the DISPLAY IEs for transmission via the D-channel.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a block diagram of apparatus for sending user data of an interactive voice response unit in accordance with one embodiment of the invention.

FIG. **2** is a perspective view of a prior art ISDN display phone for use in receiving messages from the apparatus of FIG. **1**.

FIG. **3** is a process flow of steps for utilizing the apparatus of FIG. **1**.

FIG. **4** is a block diagram of an apparatus for sending user data from a fax server in accordance with a second embodiment of the invention.

FIG. **5** is a block diagram of an apparatus for sending user data from a voicemail server in accordance with a third embodiment of the invention.

FIG. **6** is a process flow of steps for utilizing the apparatus of FIG. **4** or the apparatus of FIG. **5**.

DETAILED DESCRIPTION

With reference to FIG. **1**, an interactive voice response (IVR) unit **10** includes both voice prompt capability and visual display capability with respect to the presentation of menu options of a decision tree. Consequently, a caller at a display phone **12** may receive visual prompts, audio prompts, or both. In response, the caller identifies the selection by depression of a particular button on the telephone keypad. The IVR unit includes receivers that detect which keypad button has been depressed by a user at a remote telephone. Signal exchange by means of keypad manipulation is well known in both digital and analog telecommunications; e.g., dual tone multifrequency (DTMF) detection in an analog environment, and keypad IEs (Information Elements) in a digital environment.

The IVR unit **10** includes a conventional voice menu **14** that stores and replays voice information relating to possible menu choices. At a first level of the decision tree, an IVR unit for selecting movies may present menu options of the types of available movies, e.g. "suspense" or "comedy." The caller at the display phone **12** selects the type by depression of the appropriate button on the keypad. Based upon the selected type, the second level presents particular movies that are available.

In FIG. **1**, the voice information is transmitted to the display phone **12** via one or more B-channels **16** of an ISDN interface **18**. However, the use of ISDN telecommunications is not critical to the invention. Other links which include one or more user data channels in addition to a signaling channel may be substituted for the ISDN interface **18**. The signaling

channel in FIG. **1** is a D-channel **20** that is utilized in the conventional manner to transmit connectivity-related messages, such as CONNECT, ACKNOWLEDGE, DISCONNECT, and RELEASE signals. In addition to the exchange of connectivity-related signals that are transparent to the caller, the D-channel is conventionally used to transmit connectivity-related signals that are apparent to the caller by presentation at a display **22** of the telephone **12**. Such signals are DISPLAY IE messages or a container (i.e., envelope) having user-to-user information, and may be used to identify a party calling the display phone. In further addition to the signaling information that is conventionally exchanged via a signaling channel, such as the D-channel, the present invention employs the signaling channel to transmit user data that is typically reserved for user data channels, such as the B-channel **16**. In the embodiment of FIG. **1**, the user data information that is transmitted over the D-channel is character strings indicative of the menu options of the IVR unit **10**.

The IVR unit **10** includes a display menu **24** that generates the character strings that are associated with the menu options. In the preferred embodiment, the voice prompts and display prompts are presented simultaneously. However, this is not critical. In fact, the audio presentation of the IVR information is not required. The character string may be a sequence of ASCII characters embedded in DISPLAY IEs or user-to-user IEs in the D-channel **20**. In the embodiment of FIG. **1**, a string management device **26** is included. The device may be used to regulate the option presentation. For example, a caller at the display phone **12** may be allowed to selectively scroll through the available options by manipulating the display phone to transmit signals to the IVR unit **10**. Typically, the string management device **26** is a component of the IVR unit. Information exchanged between the IVR unit and the display phone **12** passes through a public switched telephone network (PSTN) **28** and/or private branch exchange (PBX), as is well known in the art.

FIG. **2** illustrates the features of the ISDN display phone **12**. The telephone includes the small-screen display **22** for presenting alphanumeric characters. The display allows the presentation of different options in a one-by-one time sequence. While not critical, the phone may include a first control button **32** that allows a user to scroll through the options. Pressure at one side of the control button **32** will cause a forward scrolling through the available options, while pressure at the opposite side of the control button will allow a user to scroll in reverse order. The indication of a selection of one of the options may be provided by depression of a second control button **34**. However, the conventional method of selecting among options presented by an IVR unit is to depress designated buttons of the keypad. For example, depression of the "1" key **36** will indicate a different option than depression of the "3" key **38**.

A simplified method of implementing the IVR embodiment is shown in FIG. **3**. A caller at the ISDN display phone **12** of FIGS. **1** and **2** initiates connectivity with the IVR unit **10** via the PSTN **28**, as shown at step **40**. After the call has been setup, the IVR unit transmits the available options. In the preferred embodiment, the voice menu **14** transmits the options over the B-channel **16** of the interface **18** simultaneously with the character strings from the display menu **24**. Each menu option is associated with a character string to be sent out in a DISPLAY message, as shown at steps **42** and **44**. As previously noted, the DISPLAY message may be in the form of a user-to-user container or a DISPLAY IE. Visual prompts are presented to the caller in the conventional display **22** of the ISDN phone **12**. In step **46**, the IVR

monitors the selections made by the caller and processes the selections using known techniques.

One advantage of the use of visual prompts utilizing readily available telephones **12** is that no expensive and specialized equipment is required at the IVR unit **10** in order to allow access by the hearing impaired. Because ISDN display phones are used to access the information by implementation of BRI/PRI connectivity, telecommunication devices for the deaf (TDD) are not required. Preferably, the visual prompts are transmitted regardless of the hearing capability of the caller. The use of visual prompts has potential benefits to all callers, particularly if the menu options are complex.

While the ISDN phone **12** has been described and illustrated as having the display **22** integrally formed with the housing that connects to the handset **48**, this is not critical. Optionally, the display phone may be a phone emulation device, such as a personal computer that is software driven to display telecommunication messages received via a D-channel.

FIG. **4** illustrates another embodiment of the invention. In this embodiment, the display of an ISDN phone is used to present facsimile documents from a fax server **50**. The fax server is connected to a B-channel **52** of an ISDN interface **54** to allow conventional transmission of facsimile documents to a facsimile machine or a personal computer having modem capability. However, if the apparatus detects that the person who desires access to a document is located at a display phone, a D-channel **56** of the interface is utilized to transmit the documents from a fax message storage **51**. A character recognition (CR) module **58** can be used to convert the document to character strings that are embedded into DISPLAY messages to a display phone of the type shown in FIG. **2**. Preferably, the transmission presents the document in realtime and in a line-by-line fashion, allowing the caller to scroll through the document. A string management module **60** may be responsive to signals from the ISDN phone such that the caller determines the pace of the presentation of the character strings. In FIG. **2**, the first control button **32** may be used to scroll forwardly or rearwardly through the document.

An equivalent operation may take place with respect to retrieving voicemail messages. Referring now to FIG. **5**, a voicemail server **62** having message storage **61** is connected to a B-channel **64** of an ISDN interface **66** to accommodate message retrieval in the conventional manner. Additionally, the voicemail server is connected to the D-channel **68** of the interface by means of a speech recognition module **70** and a string management module **72**. When the apparatus of FIG. **5** identifies a calling party as being at a display phone, the messages from the voicemail server may be sent to the speech recognition module **70** and the string management module **72** to transmit the messages either simultaneously with the voice transmission or independently of the voice transmission. The speech recognition module converts the recorded voicemail message into text form, such as ASCII characters, while the string management module embeds the resulting text information into DISPLAY messages compatible with transmission over the D-channel. Optionally, multimedia messages are stored at the voicemail server or other voice processing server (e.g., an IVR), so that there are associated speech and text files. The conventional speech file is transmitted via the B-channel **64** for synchronized presentation of the text that is transmitted via the D-channel **68**.

FIG. **6** illustrates process steps for implementing the fax-sending and voicemail-sending techniques for FIGS. **4** and **5**. In step **74**, connectivity is established. Typically, this is a call from a display phone, such as the one shown in FIG. **2** or a soft phone, to the facility which includes the appropriate server **50** and **62**. When the server detects that the caller is at a display phone, the CR module **58** or the speech recognition module **70** is activated.

Optionally, step **76** is included to present a menu of available messages. For example, header information of each of a number of different voicemail messages may be transmitted via the D-channel **68** to the display phone. The header information may identify the subject of the message, the author of the message, or both. If the implementation of the message-sending process includes step **76**, the server is responsive to a selection by the caller. Step **77** is a step of monitoring the connection for the caller's response. At step **78**, the appropriate response is implemented. Based upon the header information that is displayed at the display phone, the caller may select among the available messages for a full-text display.

In step **80**, the selected message is converted to character strings that are in the format for transmission via the D-channel. The character strings are embedded into DISPLAY messages in step **82**. The DISPLAY messages are transmitted in step **84** to the display phone for visual presentation to the caller.

An advantage of the facsimile embodiment of FIG. **4** is that facsimile documents can be retrieved from a facsimile server **50** without requiring access to a facsimile machine or a personal computer having modem capability. A conventional display phone may be utilized. Another advantage is that the apparatus may be used to allow a caller to scroll through a fax document without requiring the document to be fully downloaded to internal memory at the location of the calling party.

An advantage of the voicemail embodiment of FIG. **5** is that voicemail messages may be simultaneously presented audibly and visually. This may be particularly important to the hearing impaired and to a party receiving a voicemail message in a language that is not the native language of the party.

While the embodiments of FIGS. **1**, **4** and **5** are shown as being implemented separately, this is not critical. Optionally, the fax server **50**, the voicemail server **62**, and the IVR unit **10** are housed within a single message center. The use of ISDN connectivity is not critical, since other communication techniques that allow DISPLAY messages to be carried along a signaling channel that is separate from the user data channel or channels may be employed in the manner described above.

What is claimed is:

1. A telecommunications method of sending user data during a call between a source and a remote phone terminal having a display, said method comprising steps of:

establishing connectivity between said source and said phone terminal utilizing a telecommunications line having at least one digital user data channel and having a digital signaling channel, including exchanging call connectivity messages via said signaling channel;

transmitting said user data via said user data channel;

converting selected user data at said source into a format compatible with transmission via said signaling channel for visual presentation of said selected user data at said display of said remote phone terminal, said selected user data being at least a portion of said user data transmitted via said user data channel; and

transmitting said converted user data from said source utilizing said signaling channel, thereby providing dual

transmissions of at least said portion of said user data transmitted via said user data channel.

2. The method of claim 1 wherein said step of converting said selected user data includes forming digital character strings representative of menu options of an interactive voice response unit, said character strings being formatted for visual presentation at said display and wherein said step of transmitting said user data via said user data channel includes formatting said user data for audible presentation at said remote phone terminal.

3. The method of claim 1 wherein said step of converting said selected user data includes forming digital data representative of facsimile messages stored at said source, said digital data being transmitted via said signaling channel.

4. The method of claim 3 wherein said step of forming said digital data representative of facsimile messages includes utilizing character recognition processing to convert said facsimile messages.

5. The method of claim 1 wherein said step of converting said selected user data includes forming digital data representative of a voicemail message, said digital data being transmitted via said signaling channel.

6. The method of claim 5 wherein said step of forming said digital data includes utilizing speech recognition processing to convert said voicemail message into said format compatible with transmission for visual presentation at said display of said remote phone terminal.

7. The method of claim 1 wherein said step of establishing connectivity includes utilizing an ISDN line, said signaling channel being a D-channel.

8. The method of claim 7 wherein said step of converting said selected user data includes forming DISPLAY information elements (IEs) for transmission via said D-channel.

9. The method of claim 3 further comprising a step of sending a menu of facsimile messages accessible at said source for visual presentation at said display, including transmitting said menu in a format compatible with visual presentation at said display, said step of transmitting said converted user data being responsive to an input from said remote phone terminal such that said input is indicative of a selected facsimile message from said menu.

10. The method of claim 5 further comprising a step of sending a menu of voicemail messages accessible at said source for visual presentation at said display, including transmitting said menu in a format compatible with visual presentation at said display, said step of transmitting said converted user data being responsive to an input from said remote phone terminal such that said input is indicative of a selected voicemail message from said menu.

11. The method of claim 1 further comprising a step of storing a speech file indicative of a message of a voice processing server, said selected user data that are converted and transmitted utilizing said signaling channel being a text file which is associated with said speech file, said step of transmitting said converted user data being synchronized with transmitting said speech file via one of said digital user data channels.

12. A method of transmitting ISDN B-channel user data to an ISDN display phone utilizing a connection having at least one B-channel for exchanging said user data and having a D-channel for exchanging protocol messages, said method comprising steps of:

formatting a subset of said user data into a plurality of text-related character strings indicative of said subset of said user data, said subset of said user data being unrelated to ISDN protocol messages;

tagging each of said text-related character strings as DISPLAY messages for transmission via said D-channel;

displaying said subset of said user data at said ISDN display phone by sending said DISPLAY messages; and

transmitting said user data over one of said B-channels in the absence of formatting said user data into said plurality of said text-related character strings so that said subset of said user data is transmitted over both said D-channel and said one of said B-channels.

13. The method of claim 12 wherein said step of formatting said subset of said user data includes converting one of a voicemail message or a menu tree of an interactive voice response unit into a sequence of said text-related character strings for display at said ISDN display phone by transmission across said D-channel.

14. The method of claim 13 further comprising a step of transmitting speech information across at least one of said B-channels substantially simultaneously with transmission of said text-related character strings via said D-channel, thereby enabling substantially simultaneous audio and visual presentation of said user data.

15. The method of claim 12 wherein said step of formatting said subset of said user data includes converting a facsimile message into a sequence of said text-related character strings for display at said ISDN display phone.

16. An apparatus for sending messages to remote ISDN display phones comprising:

an interface to an ISDN, said interface providing a plurality of B-channels for exchanging user data and a D-channel for exchanging ISDN protocol messages, including DISPLAY messages to which said ISDN display phones are responsive;

memory having stored messages, said stored messages being user data including facsimile messages;

means connected to said memory for converting at least portions of said user data of said stored messages which include said facsimile messages to display information signals in a format compatible with transmission via said D-channel for display at said ISDN display phones as character strings, said means for converting having an output connected to said D-channel; and

first means connected to said converting means for transmitting said display information signals via said D-channel.

17. The apparatus of claim 16 further comprising second means operatively associated with said memory for transmitting selected user data of said stored messages as voice information signals via said B-channels, said selected user data being transmitted over said D-channel as display information signals by said first means and being transmitted over said B-channel as voice information signals by said second means.

18. The apparatus of claim 16 wherein said first means for converting includes a speech recognition module and wherein said stored messages include voicemail messages.

19. The apparatus of claim 16 further comprising an interactive voice response device connected to said interface to transmit menu options in a format compatible with auditory presentation at said remote ISDN display phones, said stored messages including said menu options for visual presentation at said remote ISDN display phones.

20. The apparatus of claim 16 wherein said first means for converting forms an output of DISPLAY information element (IE) messages.

* * * * *

US006345047B1

(12) **United States Patent**
Regnier

(10) **Patent No.: US 6,345,047 B1**
(45) **Date of Patent: Feb. 5, 2002**

(54) **COMPUTER TELEPHONY ADAPTER AND METHOD**

(75) Inventor: **Jean Michel Regnier**, Laval (CA)

(73) Assignee: **Northern Telecom Limited**, Montreal (CA)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **09/096,468**

(22) Filed: **Jun. 12, 1998**

(51) **Int. Cl.**[7] **H04L 12/66**

(52) **U.S. Cl.** **370/352**; 370/351; 370/353; 370/354; 370/356

(58) **Field of Search** 370/351, 352, 370/353, 354, 355, 356, 401, 410, 422, 463, 468

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* cited by examiner

Primary Examiner—Wellington Chin
Assistant Examiner—Brenda H. Pham
(74) *Attorney, Agent, or Firm*—Angela C. de Wilton

(57) **ABSTRACT**

For use at a subscriber site with a subscriber line, for simultaneously sending a telephone call from a PSTN compatible telephone terminal on the subscriber site, and IP packets from a first computer, over the subscriber line, the adaptor has a converter for converting signals from the telephone terminal into IP packets, and a multiplexer, for sending simultaneously the IP packets representing the telephone call and those from the computer, along the subscriber line. The adaptor is also arranged to handle calls without conversion to IP packets, when the subscriber telephone line is not used for carrying IP packets. Using one subscriber line, all phones in a household can remain operational, to make and receive calls, while one or more PCs are concurrently accessing online services, without needing a second line, or special equipment to increase the bandwidth transmissible over the line.

18 Claims, 10 Drawing Sheets

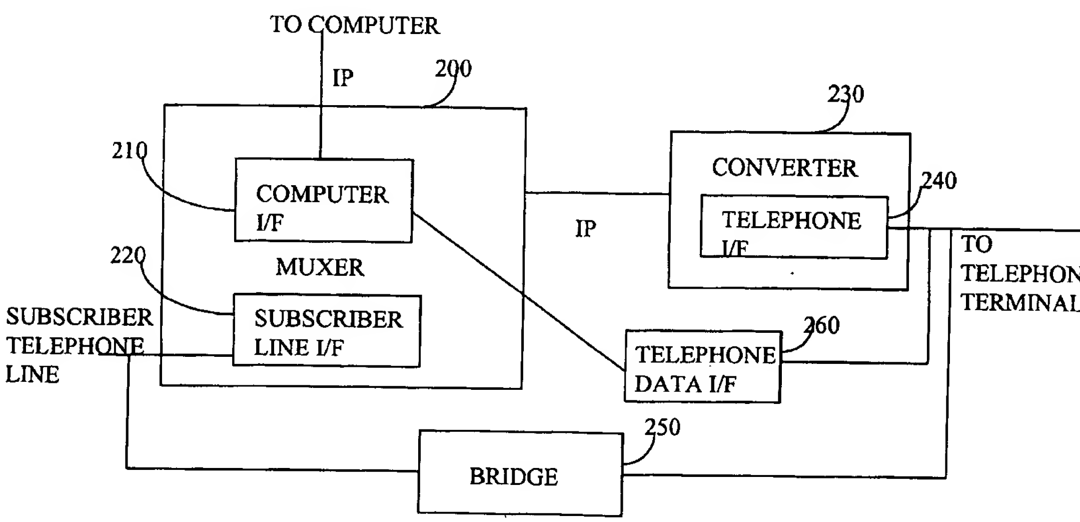


Fig 1 PRIOR ART

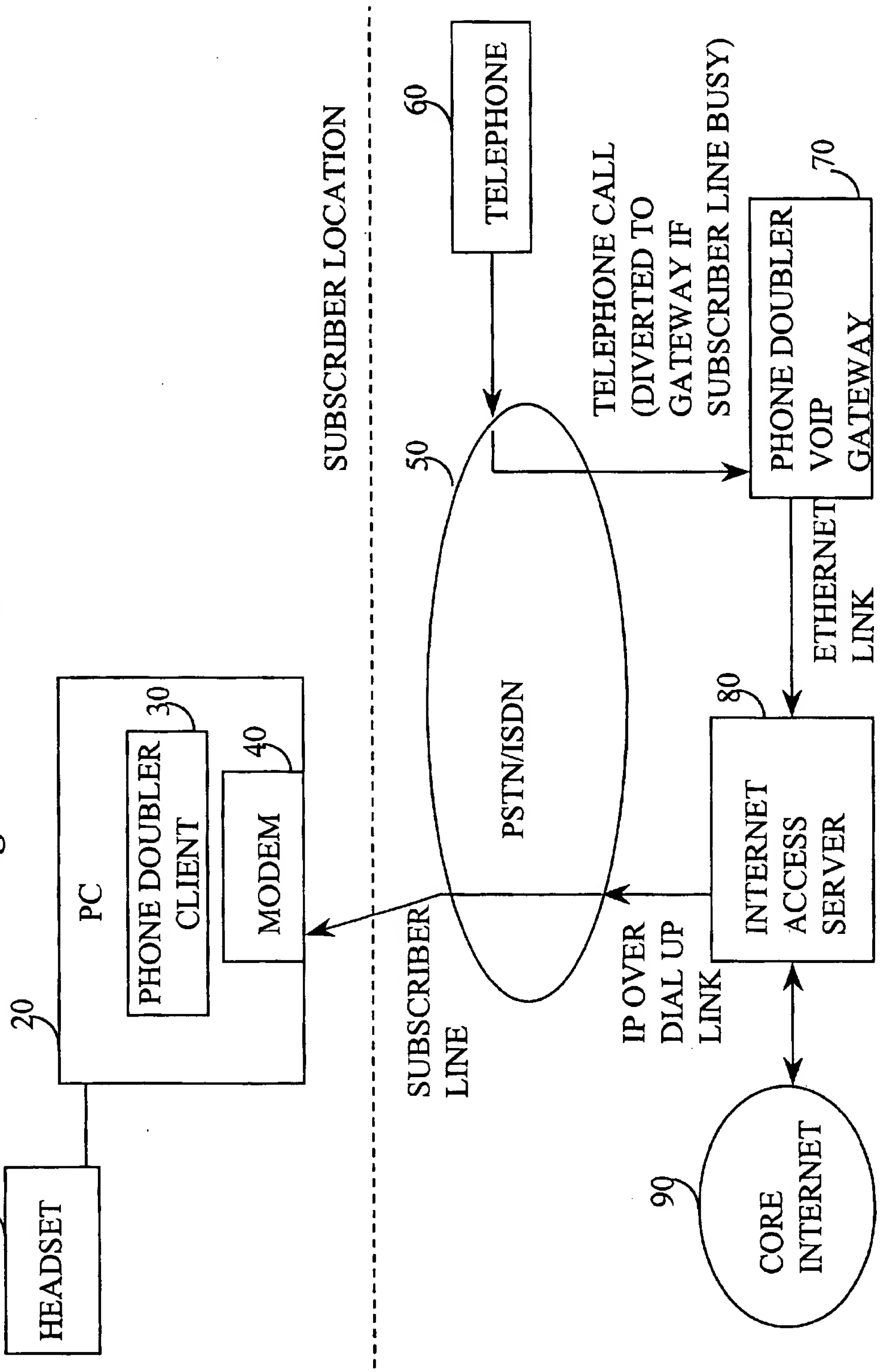
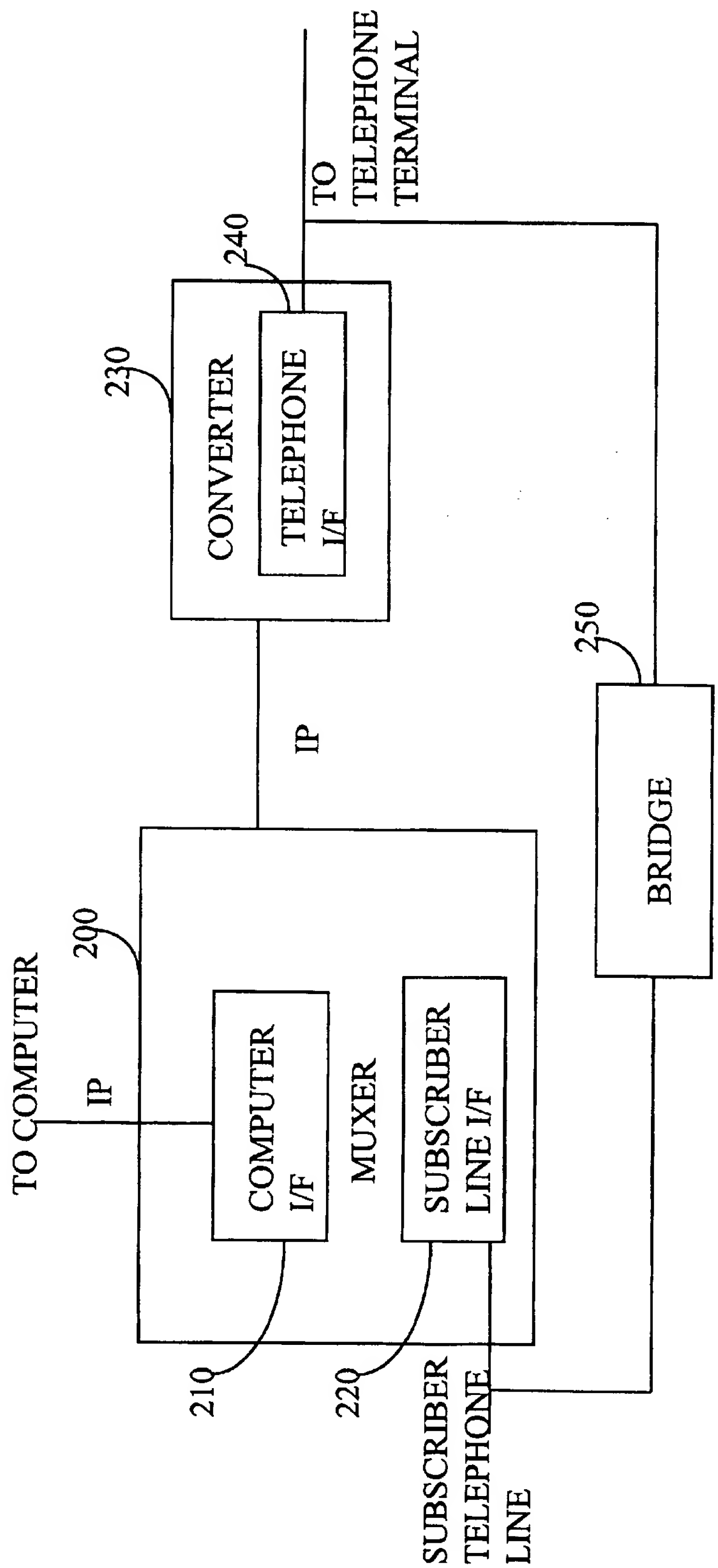


FIG. 2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240
230
CONVERTER
TELEPHONE I/F
IP
250
BRIDGE
200
TO COMPUTER
IP
COMPUTER I/F
MUXER
SUBSCRIBER LINE I/F
210
220
SUBSCRIBER TELEPHONE LINE

Fig. 3

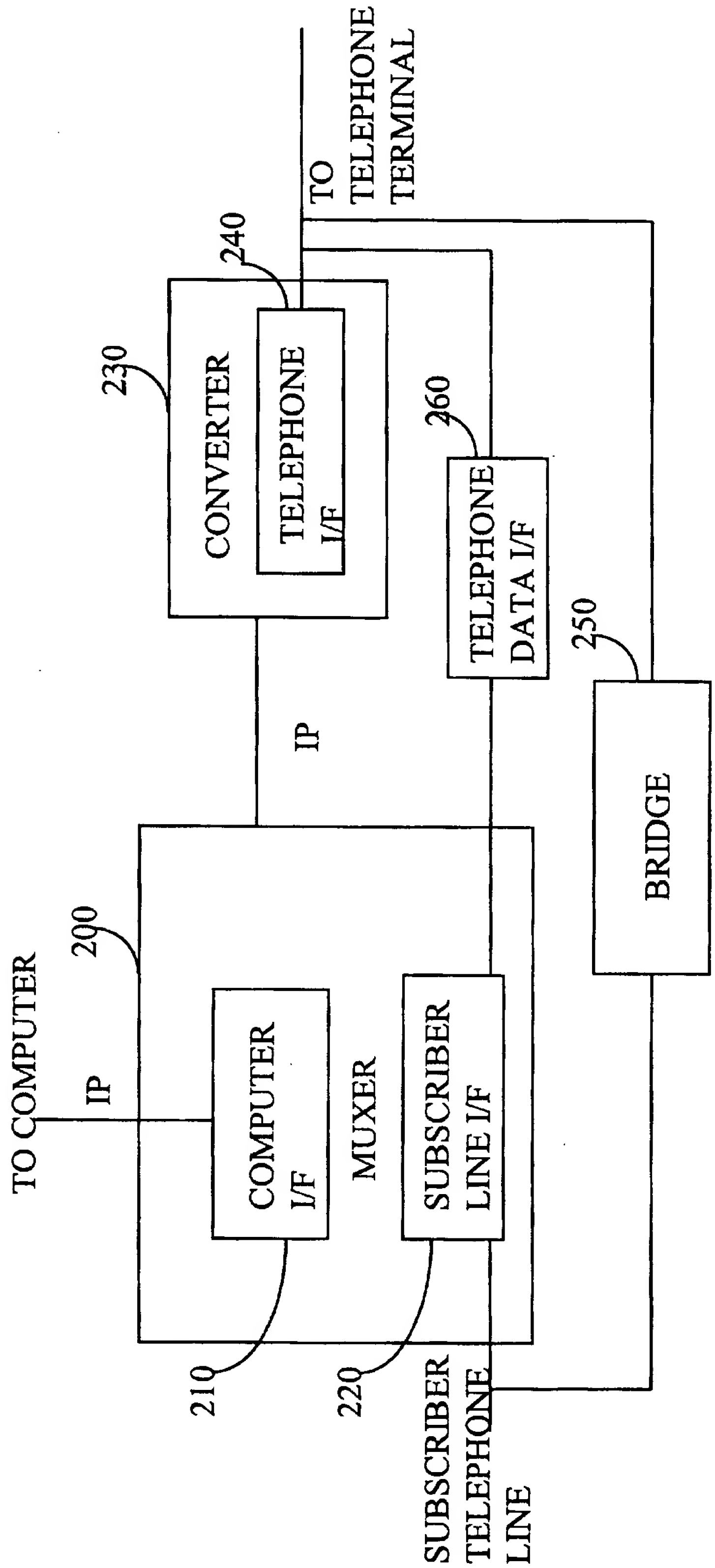

TO TELEPHONE TERMINAL
240
230
CONVERTER
TELEPHONE I/F
260
TELEPHONE DATA I/F
250
IP
BRIDGE
200
TO COMPUTER
IP
COMPUTER I/F
MUXER
SUBSCRIBER LINE I/F
210
220
SUBSCRIBER TELEPHONE LINE

Fig. 4

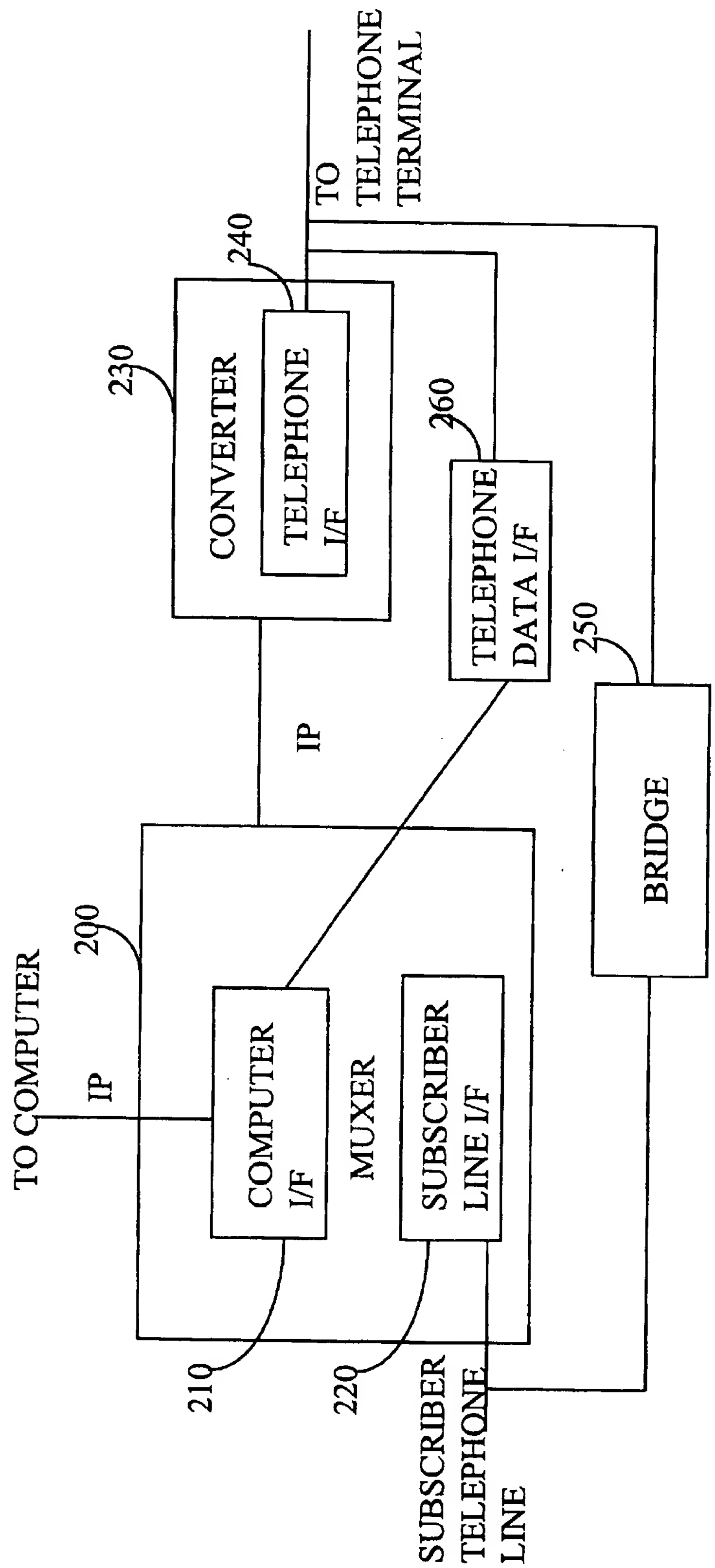

TO TELEPHONE TERMINAL
240
230
CONVERTER
TELEPHONE I/F
260
TELEPHONE DATA I/F
250
IP
BRIDGE
200
TO COMPUTER
IP
COMPUTER I/F
MUXER
SUBSCRIBER LINE I/F
210
220
SUBSCRIBER TELEPHONE LINE

Fig. 5

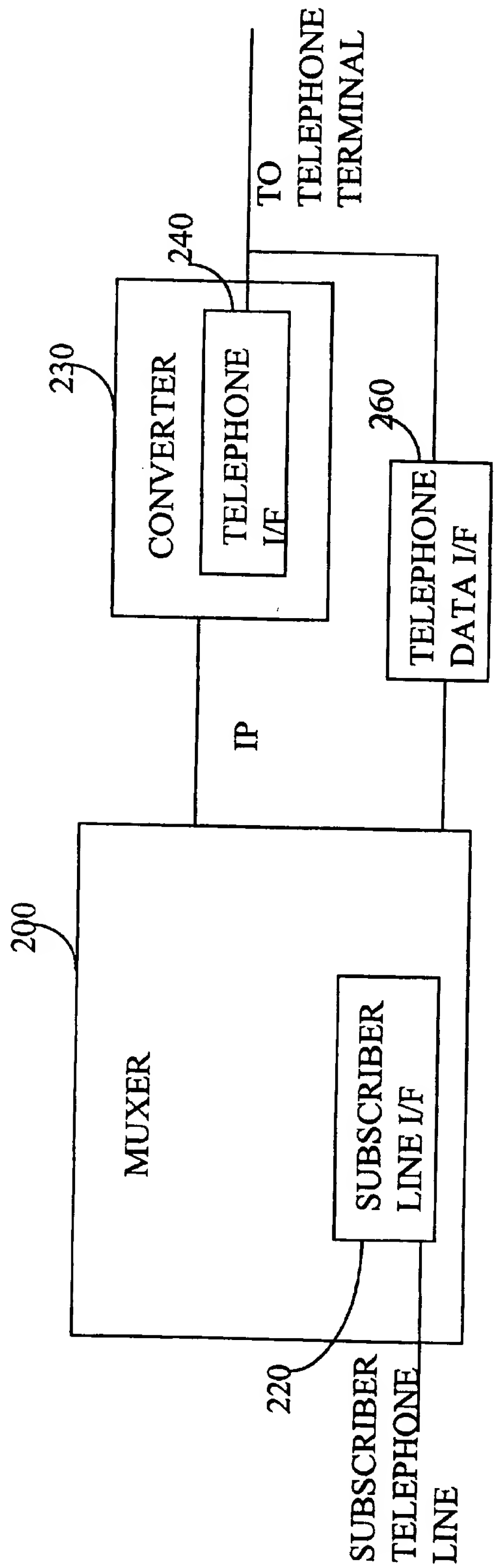
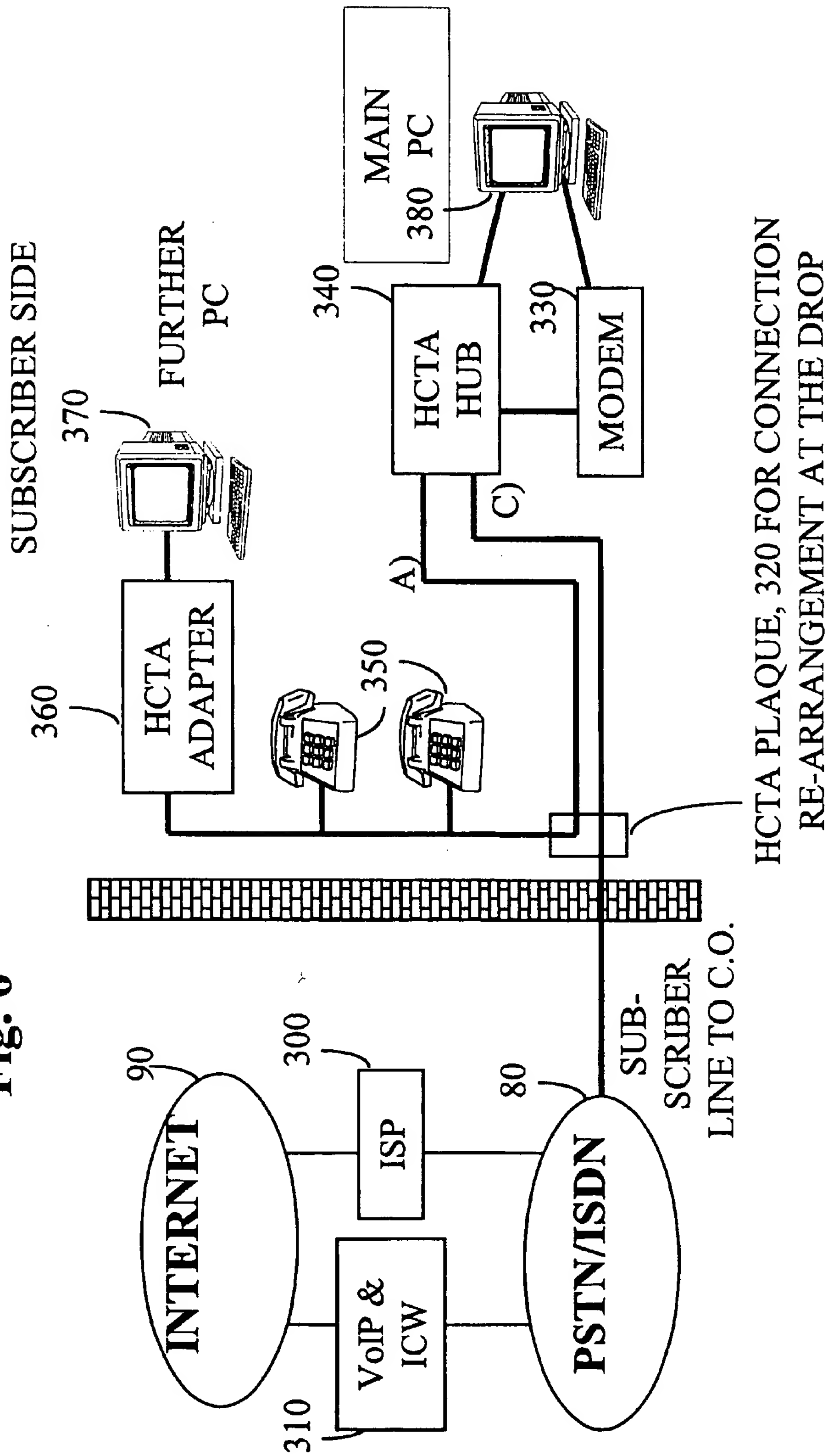

TO TELEPHONE TERMINAL
240
230
CONVERTER
TELEPHONE I/F
260
TELEPHONE DATA I/F
IP
200
MUXER
SUBSCRIBER LINE I/F
220
SUBSCRIBER TELEPHONE LINE

Fig. 6


SUBSCRIBER SIDE
FURTHER PC
370
360
HCTA ADAPTER
350
MAIN PC
380
340
HCTA HUB
330
MODEM
A)
C)
HCTA PLAQUE, 320 FOR CONNECTION
RE-ARRANGEMENT AT THE DROP
90
INTERNET
300
ISP
310
VoIP & ICW
80
PSTN/ISDN
SUB-
SCRIBER
LINE TO C.O.

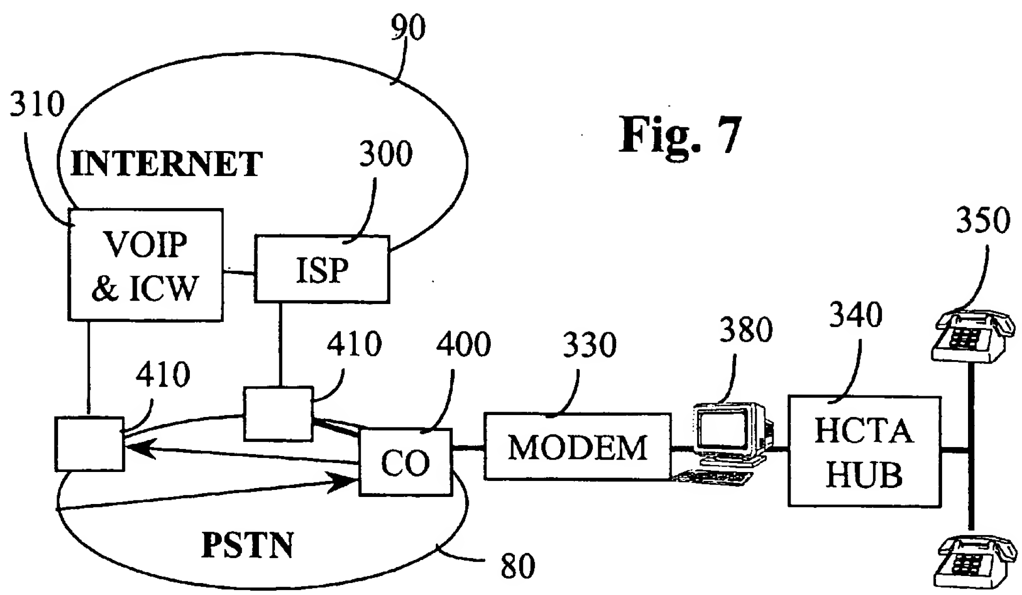

90
310
INTERNET
300
VOIP & ICW
ISP
350
410
400
330
380
340
410
CO
MODEM
HCTA HUB
PSTN
80

Fig. 7

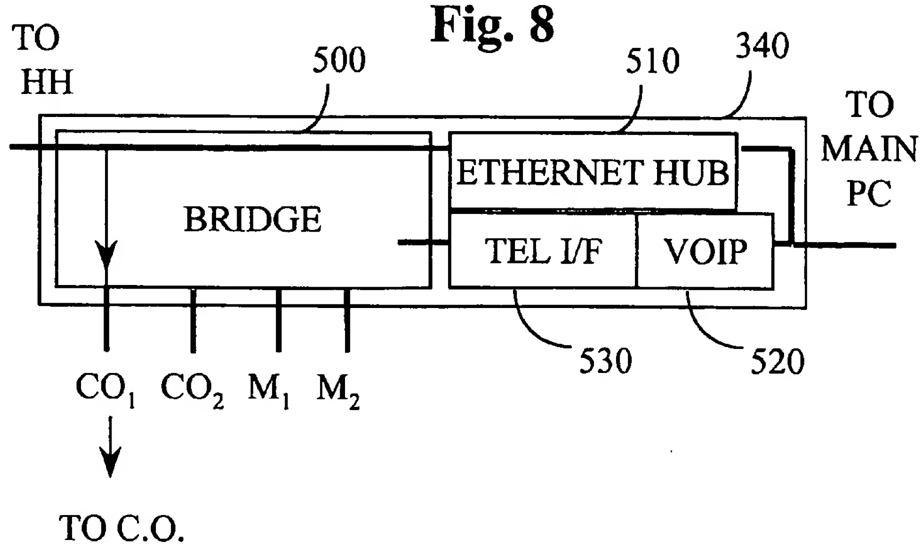

TO HH
500
510
340
TO MAIN PC
ETHERNET HUB
BRIDGE
TEL I/F
VOIP
530
520
CO_1 CO_2 M_1 M_2
TO C.O.

Fig. 8

Fig. 9

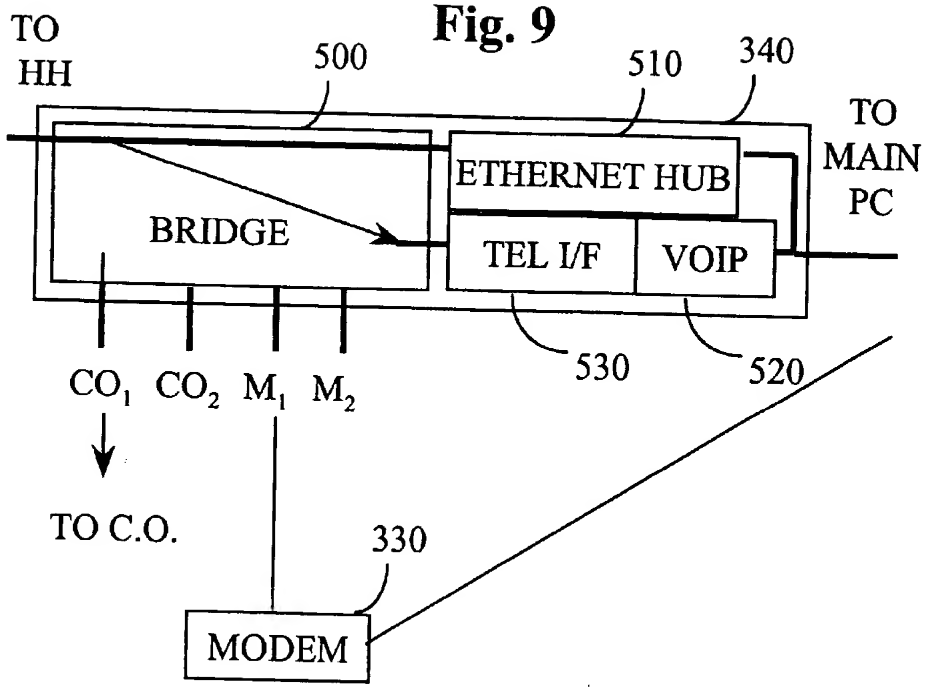

TO
HH
500
510
340
TO
MAIN
PC
ETHERNET HUB
BRIDGE
TEL I/F
VOIP
530
520
CO_1
CO_2
M_1
M_2
TO C.O.
330
MODEM

Fig. 10

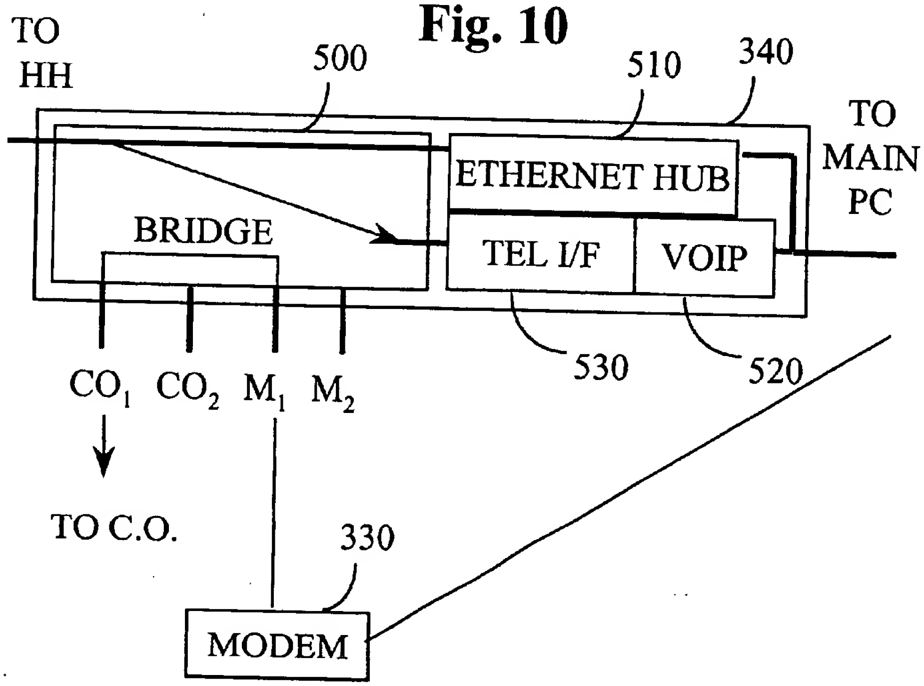

TO
HH
500
510
340
TO
MAIN
PC
ETHERNET HUB
BRIDGE
TEL I/F
VOIP
530
520
CO_1
CO_2
M_1
M_2
TO C.O.
330
MODEM

Fig. 11

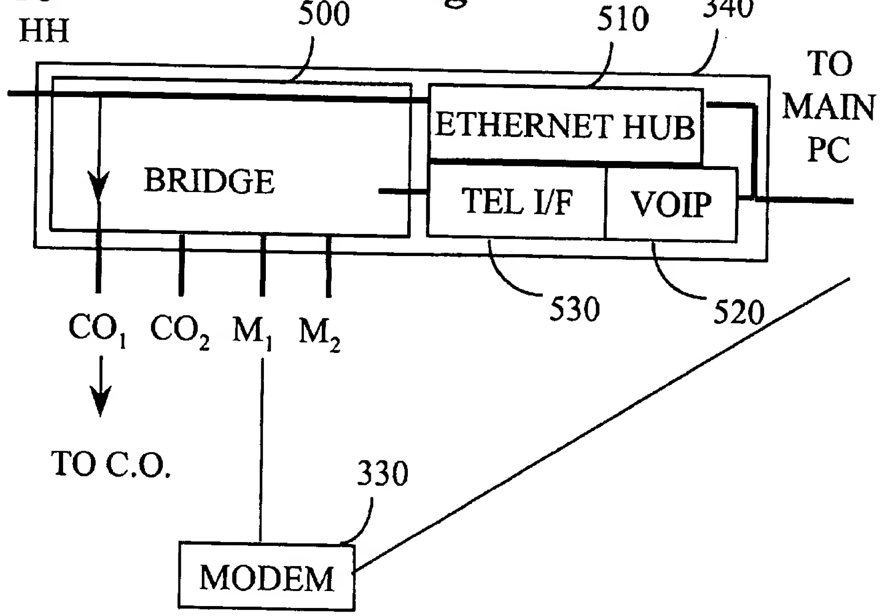

TO
HH
500
510
340
TO
MAIN
PC
ETHERNET HUB
BRIDGE
TEL I/F
VOIP
530
520
CO1
CO2
M1
M2
TO C.O.
330
MODEM

Fig. 12

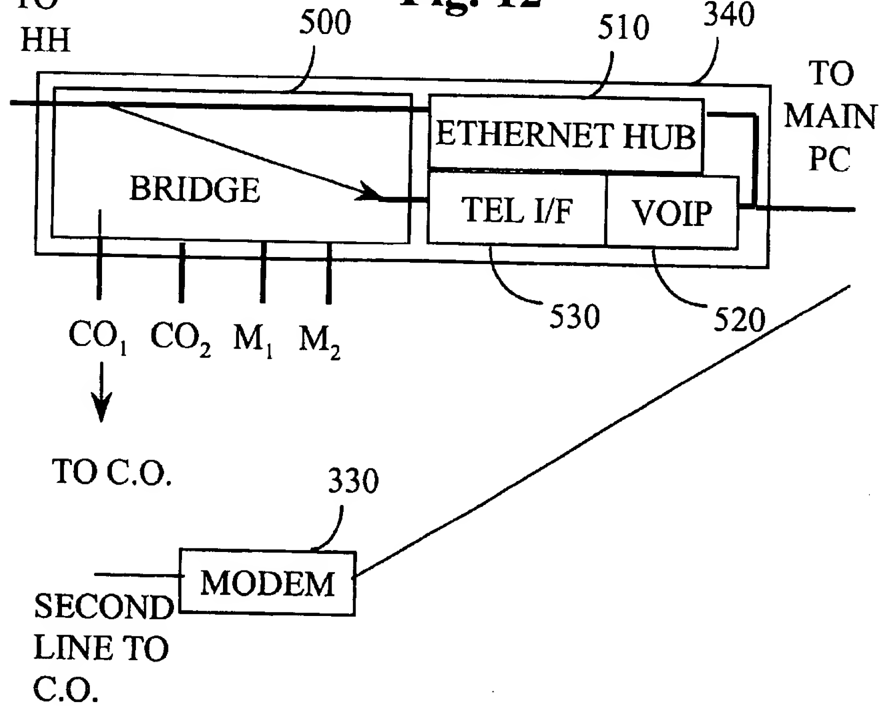

TO
HH
500
510
340
TO
MAIN
PC
ETHERNET HUB
BRIDGE
TEL I/F
VOIP
530
520
CO1
CO2
M1
M2
TO C.O.
330
MODEM
SECOND
LINE TO
C.O.

Fig. 13

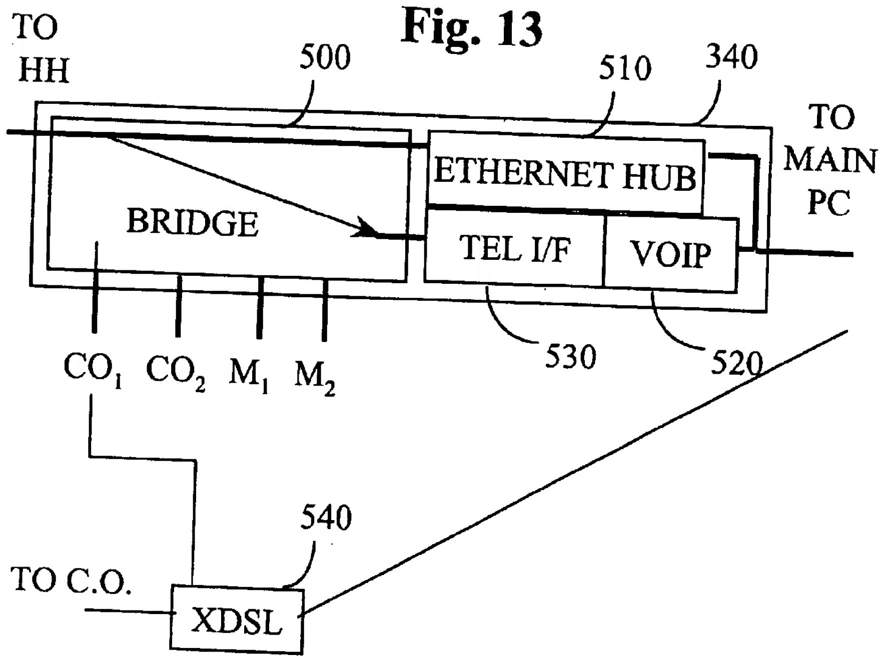
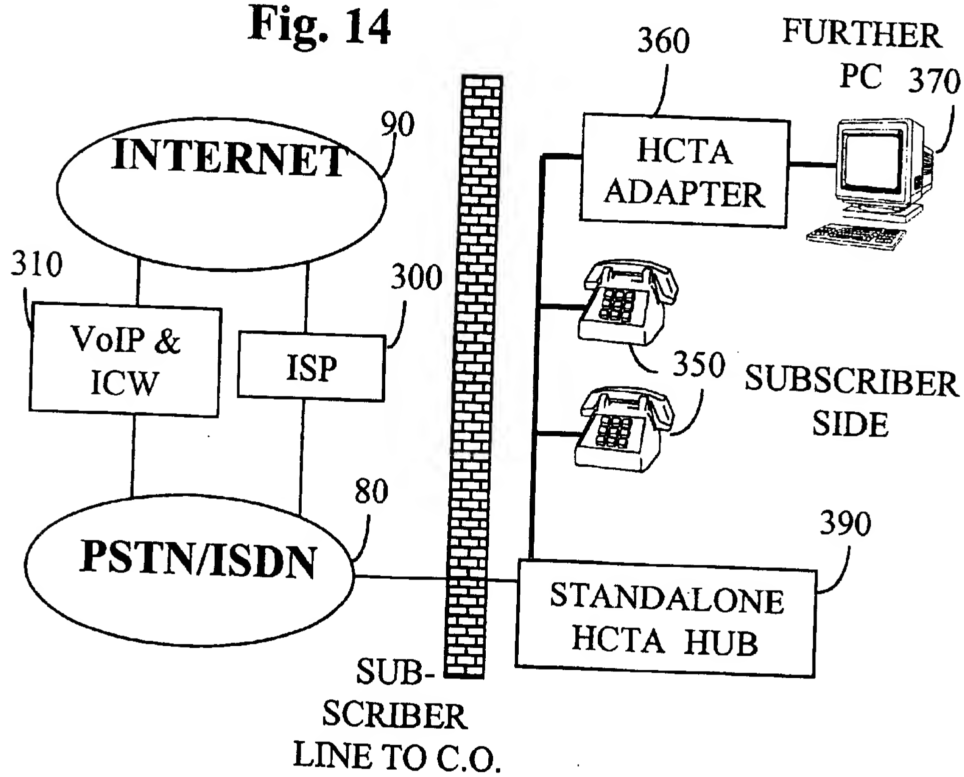

TO HH
500
510
340
TO MAIN PC
ETHERNET HUB
BRIDGE
TEL I/F
VOIP
530
520
CO_1 CO_2 M_1 M_2
540
TO C.O.
XDSL

Fig. 14


360
FURTHER PC 370
INTERNET
90
HCTA ADAPTER
310
300
VoIP & ICW
ISP
350
SUBSCRIBER SIDE
80
390
PSTN/ISDN
STANDALONE HCTA HUB
SUB-SCRIBER LINE TO C.O.

COMPUTER TELEPHONY ADAPTER AND METHOD

BACKGROUND TO THE INVENTION

1. Field of the invention

The invention relates to apparatus for simultaneously sending a telephone call over a subscriber telephone line from a PSTN compatible telephone terminal on a subscriber site, and IP packets from a first computer, to apparatus for simultaneously receiving a telephone call to a PSTN compatible telephone terminal on the subscriber site, and IP packets directed to a first computer, both sent over a subscriber telephone line, to methods of using such apparatus, and to software for carrying out such methods.

2. Background Art

The most common way of accessing on-line services using the Internet is via a modem link over a dial-up PSTN connection. However, this may tie up the userls telephone line for long periods. Incoming and outgoing voice telephone calls cannot be made or completed. Similarly, when the telephone line is being used for a voice telephone call, the Internet cannot be accessed.

One known solution to this problem is to have a second line installed. However, additional expense is involved and the second line may be accessible from only one room in the house or small office, unless major rewiring work is carried out to make both lines accessible in many rooms.

An alternative solution is called the Internet Phonejack. This is a telephony expansion card which can be plugged into a PC, and enables Internet telephony applications such as Microsoft Netmeeting to interface with conventional telephone terminals. The card emulates an ordinary subscriber telephone line, and thus enables such telephone terminals to be used for voice calls at the same time as the computer user is accessing the Internet. The analog voice signals are converted into Internet Protocol (IP) packets which can be interleaved with IP packets used by the computer for accessing other services over the Internet simultaneously.

Another known system is called the Phone Doubler, produced by Ericsson. FIG. **1** illustrates this known arrangement in schematic form. A userls PC (Personal Computer) **20** is connected to a subscriber telephone line using a modem **40**. Phone doubler client software **30** runs on the PC. A headset **10** with microphone and earphone is connected to the PC.

Away from the subscriberls location, the rest of the PSTN/ISDN 50 links other elements of the arrangement. A VOIP (Voice over Internet Protocol) gateway **70** is connected to the PSTN and to an Internet access server **80**. The VOIP gateway and the Internet access server may be co-located and connected by a local area network such as an ethernet link. The Internet access server is linked to the rest of the Internet **90**, and also to the PSTN.

In operation, when an incoming call from a remote telephone **60** is routed by the PSTN to the subscriber with the PC **20**, if the subscriber is already connected to the Internet over his subscriber line, the local central office (not shown) in the PSTN will be arranged to divert the call to the phone doubler VOIP gateway, assuming the subscriber has previously subscribed to this phone doubler service. The VOIP gateway will receive the call and will look up the IP address of the subscriber, based on the telephone number of the subscriber which has been dialed from the remote telephone **60**. The gateway determines if the user already has an IP session with the Internet access server, and if so, communicates with the phone doubler client software **30** to offer the user the option of taking the call.

If the user accepts the call, the gateway converts the incoming voice signals into UDP (User Datagram Protocol) format, which uses IP for transmission. The phone doubler client **30** receives the IP packets, assembles the UDP datagrams, converts them back into the analog voice signals, which are fed to the headset **10**. Voice signals from the subscriber can be transmitted in a corresponding way back to the remote telephone **60**. This achieves a virtual second line for a user, to enable simultaneous voice and on-line access over a single subscriber line.

It is also known to provide systems to enable multiple computers to access the Internet over a single subscriber line simultaneously. The multiple computers may be connected over a local area network (LAN), and an Internet LAN bridge can be used to enable multiple LAN users to share a single dial-up Internet connection at the same time.

Another known system, called HomeRun has been proposed by TUT Systems, and uses existing telephone wiring in the home or office to create an ethernet LAN. It has been proposed to use this to provide Internet access throughout the home or office by linking all the computers to a modem which can maintain a single dial up connection to an ISP (Internet Service Provider), by creating a LAN over the existing telephone wiring. The networking signals and standard telephone service can coexist on the same wires at the same time, using frequency division multiplexing. Ordinary telephones can be used without alteration. PCs can be plugged in to the telephone sockets using interface cards which support a standard RJ11 phone jack, and include an ethernet interface.

It is also known to modify the equipment at each end of the subscriber telephone line to provide a higher bandwidth service over the subscriber line, e.g. 128 kbps for ISDN (Integrated Services Digital Network) or higher for xDSL (High Speed Digital Subscriber Line) type systems. For an ISDN line, two channels are multiplexed over a single physical path. The interfaces at each end may be arranged so that voice calls use one of the two channels, while the other may be used to carry data, to enable online services to be accessed simultaneously. xDSL encompasses a range of different types of digital subscriber line systems including ADSL (Asynchronous Digital Subscriber Line), requiring specialized equipment at both ends of the subscriber line.

In this document, PSTN compatible telephone terminals shall be defined as excluding ISDN or ADSL compatibility. PSTN compatible is intended to encompass compatibility with a basic public telephone service interface. Such terminals use a conventional tip and ring analog interface, for use with one 64 kbit channel in the PSTN network. They may have a wired handset or a cordless handset, or a speaker phone, for example.

The term IP is intended to encompass any version of the Internet Protocol and any other protocols which may be used for carrying Internet traffic with ISO layer three functions.

SUMMARY OF THE INVENTION

It is an object of the invention to provide improved methods and apparatus.

According to a first aspect of the invention there is provided apparatus for use with a subscriber telephone line connecting a subscriber site and a telephone network, for simultaneously sending a telephone call from a PSTN compatible telephone terminal on the subscriber site, and IP packets from a first computer, over the subscriber telephone line, the apparatus comprising:

a converter having a PSTN compatible telephone line interface for coupling to the PSTN compatible telephone terminal, the converter being for converting signals from the PSTN compatible telephone terminal into IP packets representing the telephone call; and

a multiplexer coupled to the converter, and having a computer interface for coupling to the first computer, and a subscriber line interface for coupling to the subscriber telephone line, the multiplexer being arranged for sending simultaneously the IP packets representing the telephone call and those from the computer, along the subscriber telephone line;

the apparatus being arranged to handle telephone calls without conversion to IP packets, between the telephone terminal and the subscriber telephone line when the subscriber telephone line is not used for carrying IP packets.

One of the problems of the prior art, discovered and addressed by this aspect of the invention is the ability to provide, using one subscriber line, e.g. a conventional POTS (Plain Old Telephone Service) line, that all phones in a household remain operational (i.e., they can be used to make and receive calls), while one or more PCs are concurrently accessing online services, without needing a second line, or special equipment to increase the bandwidth transmissible over the line by a factor of two or more.

One advantage of providing the capability of handling phone calls from the PSTN compatible telephone terminal on the subscriber line either over a telephone interface, or in the form of IP packets is that it enables a single subscriber line to be used for both purposes. This cannot be achieved with traditional telephones which can always only use the telephone interface, nor with the Internet phone jack and Phone doubler products which can always only carry calls in the form of IP packets.

Another advantage arises from the converter having a PSTN compatible interface. One consequence of this is that existing telephones, extension telephones and telephone cabling in a house or office can continue to be used even when the subscriber telephone line is carrying IP packets. This previously would have required a second line. In the above mentioned Internet phone jack and Phone Doubler products, telephony is limited to a headset and microphone belonging to the computer, or to a telephone terminal belonging to the computer.

With respect to the above mentioned known ISDN or ADSL systems, such systems have no converter, but require expensive dedicated linecards and other expensive equipment at the central office and the Internet service provider to provide a bandwidth of 128 Kbps or higher. The converter enables telephony and IP services to be carried simultaneously at much lower cost, over a conventional subscriber telephone line. The apparatus does not require additional equipment on the subscriber telephone line to provide a high bandwidth service. It only requires the bandwidth that a conventional subscriber telephone line supports for conventional dialed telephone calls.

Preferably the apparatus further comprises a bridge for selectively coupling the subscriber telephone line to an internal telephone line to which the telephone terminal is attached.

This can enable an advantage to be achieved in that telephone calls can be routed directly without using the converter when the computer is not using the subscriber line, and in that telephone calls can be converted into IP packets and be routed in an IP session when the computer is using the subscriber telephone line for an IP session. Further, this can also enable a second advantage in that when the internal cabling is not coupled to the subscriber telephone line, the internal cabling can be used for purposes such as telephone calls between terminals within the customer site, without interfering with signals on the subscriber telephone line.

Preferably, the telephone terminal is coupled to the apparatus via an internal telephone line, the apparatus further comprising a telephone data interface coupled to the multiplexer and arranged for coupling to the internal telephone line, for receiving data from a further computer coupled to the internal telephone line, and distinguishing this data from analog signals from the telephone terminals, the multiplexer being arranged to send the data from the further computer along the subscriber telephone line.

One advantage which can arise here is that the further computer can make use of existing internal telephone cabling in a house or office to access online services over the subscriber line. Furthermore, it may be possible to do so simultaneously with the first computer using the subscriber line, or with a telephone call taking place, or even simultaneously with both of these possibilities. The above mentioned Homerun product can be used to enable multiple computers to concurrently access online services over a conventional subscriber telephone line and existing internal cabling. However, in such an arrangement the existing telephone terminals connected to the internal cabling cannot be used to send or receive calls while a computer is online.

Preferably, the telephone terminal is coupled to the apparatus via an internal telephone line, the apparatus further comprising a telephone data interface coupled to the multiplexer and for coupling to the computer interface, for receiving data from a further computer coupled to the internal telephone line, the multiplexer being arranged to send the data from the further computer to the first computer.

This enables multiple computers to communicate amongst themselves using existing telephony cabling. This enables many of the advantages of a local area network to be achieved using existing cabling. Furthermore the combination of this local area network capability with the capability of the first computer to access online services while maintaining telephony, can make it possible for the further computer to access online services via the first computer. Alternatively, or concurrently, telephone terminals can make and receive calls.

Preferably the telephone line interface and the telephone data interface are arranged to communicate over the internal telephone line simultaneously.

The combination of this local area network capability with the capability of the apparatus to access online services while maintaining telephony, can make it possible for any or many computers to access online services via the apparatus while concurrently allowing telephone terminals anywhere on the internal telephone line to make and receive calls. In the above mentioned Homerun product, a computer can communicate with a further computer while a call is being made from a conventional telephone terminal. However, in such an arrangement, the computers may not access online services at the same time as the call is being made from the conventional telephone terminal.

Preferably the telephone line interface further comprises an internal line monitor for determining the state of telephony activity from the telephone terminal, the apparatus being arranged to control a telephone call in response to the determination.

An advantage of this is that it enables the call to be controlled as desired by the end user.

Preferably the apparatus further comprises a subscriber line monitor for determining the state of the subscriber telephone line the apparatus being arranged to control a telephone call in response to the determination.

This can help in deciding if the call can be established in conventional non-IP form.

Preferably the apparatus further comprises an IP session monitor for determining when there is an IP session already existing, the apparatus being arranged to control a telephone call in response to this determination.

This can help decide whether the call can be established using an existing IP session.

Preferably the apparatus further comprises an internal line monitor for determining the state of telephony activity from

the telephone terminal, and a subscriber line monitor for determining the state of the subscriber telephone line, the apparatus being arranged to control the call additionally on the basis of the outputs of the internal line monitor and the subscriber line monitor.

In combination the capability to monitor telephony activity from the telephone terminal and the capability to monitor the state of the subscriber telephone line, provide an advantage for example in enabling voice over IP resources to be used to establish a call when the subscriber telephone line is being used already to access an online service, or according to other criteria, e.g. if it is more cost effective. A second advantage would be to facilitate a decision to establish the call as a conventional analog telephony call for example when an IP session is not established and the subscriber telephone line is idle.

Preferably the telephone line interface is arranged to determine what number has been dialed, and establish the call as a VoIP call, or establish the call as a conventional telephone call, on the basis of the number dialed.

An advantage is that the decision of whether to route the call using VoIP or non VoIP can be made automatically for example when it is more cost effective, or when instructed by the subscriber.

Preferably the apparatus further comprises an internal line monitor, for determining the state of telephony activity from the telephone terminal, the apparatus being arranged to send a busy indication in response to an incoming call for the telephone terminal, if an IP session is established and if the internal line monitor determines that the telephone terminal is busy.

Preferably the apparatus is arranged to prompt a caller of an incoming call to indicate a party that he intends to reach, and the telephone interface is arranged to process the call on the basis of the party indicated by the caller.

Preferably the telephone line interface is arranged to output an alert specific to the party indicated by the caller.

Preferably the multiplexer further comprises a second subscriber line interface for coupling to a second subscriber line.

This can enhance the apparatus by allowing for example a non VoIP call on one line at the same time as one or more IP sessions are being handled over the other line.

Preferably the multiplexer further comprises a second subscriber line interface for handling a second connection over the subscriber telephone line, for use with a high bandwidth coupler for multiplexing the second connection over the subscriber telephone line.

This can enhance the apparatus by allowing for example a non VoIP call on the subscriber line at the same time as one or more IP sessions are being handled in the second connection over the subscriber telephone line.

According to a further aspect of the invention, there is provided apparatus for use with a subscriber telephone line connecting a subscriber site and a telephone network, for simultaneously sending a telephone call from a PSTN compatible telephone terminal on the subscriber site coupled to an internal telephone line, and data from a computer coupled to the internal telephone line, over the subscriber telephone line, the apparatus comprising:

a converter having a PSTN compatible telephone line interface for coupling to the PSTN compatible telephone terminal, the converter being for converting signals from the PSTN compatible telephone terminal into IP packets representing the telephone call;

a telephone data interface for coupling to the internal telephone line for receiving the data from the computer over the internal telephone line, the telephone line interface and the telephone data interface being arranged to communicate over the internal telephone line simultaneously; and

a multiplexer coupled to the converter, and coupled to the telephone data interface, and having a subscriber line interface for coupling to the subscriber telephone line, the multiplexer being arranged for sending simultaneously the telephone call and the data from the computer, along the subscriber telephone line in the form of IP packets.

The above mentioned problem of the prior art, is also addressed by this aspect of the invention. All phones in a household remain operational (i.e., they can be used to make and receive calls), while one or more PCs are concurrently accessing online services, without needing a second line, or special equipment to increase the bandwidth transmissible over the line by a factor of two or more. Compared to the first aspect, at least one of the PCs makes use of the internal telephone line to reach the subscriber line, but the direct computer connection with the apparatus avoiding using the internal telephone line, is optional.

Preferably the apparatus further comprises a bridge for selectively coupling the subscriber telephone line to the internal telephone line.

Preferably the telephone data interface comprises a local area network interface for coupling more than one computer to the apparatus.

According to a further aspect of the invention, there is provided apparatus for use with a subscriber telephone line connecting a subscriber site and a telephone network, for simultaneously receiving a telephone call to a PSTN compatible telephone terminal on the subscriber site, and IP packets directed to a first computer, both sent over the subscriber telephone line, the apparatus comprising:

a converter having a PSTN compatible telephone line interface for coupling to the PSTN compatible telephone terminal, the converter being for converting from IP packets representing the telephone call into signals for the PSTN compatible telephone terminal; and

a demultiplexer coupled to the converter, and having a computer interface for coupling to the first computer, and a subscriber line interface for coupling to the subscriber telephone line, the demultiplexer being arranged for receiving simultaneously the IP packets representing the telephone call and those for the computer, over the subscriber telephone line;

the apparatus being arranged to handle telephone calls without conversion to IP packets, between the telephone terminal and the subscriber telephone line when the subscriber telephone line is not used for carrying IP packets.

According to a further aspect of the invention, there is provided apparatus for use with a subscriber telephone line connecting a subscriber site and a telephone network, for simultaneously receiving over the subscriber telephone line from the subscriber site a telephone call from a PSTN compatible telephone terminal coupled to an internal telephone line and data for a computer coupled to the internal telephone line, the apparatus comprising:

a converter having a PSTN compatible telephone line interface for coupling to the PSTN compatible telephone terminal, the converter being for converting IP packets representing the telephone call into signals for the PSTN compatible telephone terminal; a telephone data interface for coupling to the internal telephone line for receiving the data from the computer over the internal telephone line; and

a demultiplexer coupled to the converter, and to the telephone data interface, and a subscriber line interface for coupling to the subscriber telephone line, the demultiplexer being arranged for receiving simulta-

neously the telephone call and the data for the computer, over the subscriber telephone line in the form of IP packets.

According to a further aspect of the invention, there is provided a method of simultaneously sending a telephone call from a PSTN compatible telephone terminal on a subscriber site and IP packets from a first computer, over a subscriber telephone line, the method comprising the steps of:

converting signals from the PSTN compatible telephone terminal into IP packets representing the telephone call;

using a multiplexer coupled to the converter, and having a computer interface for coupling to the first computer, and a subscriber line interface for coupling to the subscriber telephone line, to send simultaneously the IP packets representing the telephone call and those from the computer, along the subscriber telephone line; and handling telephone calls without conversion to IP packets, between the telephone terminal and the subscriber telephone line when the subscriber telephone line is not used for carrying IP packets.

According to a further aspect of the invention, there is provided a method of simultaneously sending a telephone call from a PSTN compatible telephone terminal on a subscriber site coupled to an internal telephone line, and data from a computer coupled to the internal telephone line, over a subscriber telephone line, the method comprising the steps of:

converting signals from the PSTN compatible telephone terminal into IP packets representing the telephone call; using a telephone data interface coupled to the internal telephone line for receiving the data from the computer over the internal telephone line; and

using a multiplexer coupled to the converter, and to the telephone data interface, and having a subscriber line interface for coupling to the subscriber telephone line, to send simultaneously the telephone call and the data from the computer, along the subscriber telephone line in the form of IP packets.

Another aspect of the invention provides software stored on a computer readable medium for carrying out the above methods.

Any of the preferred features may be combined, and combined with any aspect of the invention, as would be apparent to a person skilled in the art. Other advantages will be apparent to a person skilled in the art, particularly in relation to prior art other than that mentioned above.

To show, by way of example, how to put the invention into practice, embodiments will now be described in more detail, with reference to the accompanying drawings.

BRIEF DESCRIPTION OF DRAWINGS

FIG. 1 shows the known Phone Doubler arrangement;

FIG. 2 shows a schematic view of apparatus of a first embodiment of the invention;

FIG. 3 shows a schematic view of an alternative embodiment of the invention;

FIG. 4 shows a schematic view of apparatus according to a further embodiment of the invention;

FIG. 5 shows a schematic view of apparatus according to a yet further embodiment of the invention;

FIG. 6 shows an overview of a further embodiment which includes features corresponding to the embodiments of FIGS. 2,3 and 4;

FIG. 7 shows in more detail in schematic form the operation of network services in the embodiment of FIG. 6;

FIG. 8 shows in more detail in schematic form the HCTA hub of FIG. 6 or FIG. 7;

FIGS. 9 to 11 show the HCTA hub of FIG. 8, in various operational modes;

FIG. 12 shows the HCTA hub of FIG. 8 arranged to operate with two subscriber lines;

FIG. 13 shows the HCTA hub of FIG. 8 arranged to operate with an xDSL type of subscriber line;

FIG. 14 shows a configuration overview of a standalone embodiment using the arrangement of FIG. 5.

DETAILED DESCRIPTION

Introduction

Many online households possess multiple personal computers. For instance, parents may own the new PC, and relegate to their children the older PC, or two professionals may each have their own PCs. Currently, multiple PCs in a household cannot communicate amongst themselves unless new wiring is installed. Because they cannot communicate, multiple PCs cannot share devices (e.g., modems, printers) and cannot exchange data. They also cannot use services concurrently (e.g., an Internet account).

An arrangement of hardware and software in various embodiments which addresses such problems, will be referred to as the Home Computer Telephony Adapter (HCTA). It may be implemented as a PC card using software run by the PC, or as a peripheral external to the PC. The overall purpose is to enhance communication services for online households. Most notably, it enables telephony to be maintained for the whole household while PCs are accessing online services. The HCTA also provides a data communication capability over the existing in-house wiring. This enables PCs to exchange data and to share devices and services. For instance, HCTA enables multiple PCs to be concurrently online.

The HCTA requires its end user to subscribe to one POTS line and to subscribe to an Internet service with Voice-over-IP (VoIP) and Internet Call Waiting (ICW) features, which are commercially available, and expected to become widely available. It also requires the end user to possess a PC with a modem, preferably 56 kbps. HCTA requires no second line or special Telco (Telephone Operating Company) equipment to function properly, although performance would improve with such capabilities, and some embodiments including such capabilities are discussed below.

A given HCTA implementation may differ from existing products in various ways including the following:

a) It may operate with the in-house wiring uncoupled from the PSTN access line.

b) It may monitor telephony activities on the in-house wiring and on the PSTN access line, and PC access to online services.

c) It may provide telephony for the household over the PSTN when no PC is online and provide telephony for the household through the PC when the PC is online.

These differences enable the HCTA to offer basic Key system/LAN/Bridge capabilities to the household. Specifically, some notable features which will be explained in more detail below are:

a) Regarding voice communication:

1)To enable people to make and receive calls from any phone in the household while PCs are connected to online services.

2)To enable calls within the household (intercom), and to provide distinctive ringing to all household occupants.

3)Access to cheap Internet-based toll services from any phone and concurrent calls from PCs and phones.

b) Regarding data communication:

To enable multiple PCs in the household to exchange data, to be concurrently online and to share devices (e.g., a printer, a modem). These services are provided over the existing in-house wiring. They require no re-wiring or additional wiring. To support these services, the HCTA relies on available technology for telephony over the Internet (for example VoIP) and on available technology for multiplexing of voice and data signals over internal wiring in the household for example by frequency division multiplexing.

FIG. 2 schematic view of first embodiment of the invention

FIG. 2 shows in schematic form some of the features of one embodiment of the apparatus at the subscriber side, for use with the PSTN, and the Internet as shown in FIG. 1. The subscriber side apparatus includes a multiplexer 200, a computer interface 210, and a subscriber line interface 220. The multiplexer is coupled to a converter 230. The converter feeds a telephone interface 240, coupled to a conventional PSTN telephone terminal (not shown).

A bridge 250 enables the subscriber telephone line to be coupled to the PSTN telephone terminal without going through the multiplexer 200 and the converter 230. The apparatus is capable of sending a telephone call from the PSTN telephone terminal, and IP packets from the computer, over the subscriber telephone line. The converter is for converting signals from the PSTN telephone terminal into IP packets representing the telephone call. The apparatus may additionally handle telephone calls without conversion to IP packets, between the telephone terminal and the subscriber telephone line, when this subscriber telephone line is not being used for carrying IP packets.

The apparatus may be arranged for outgoing calls and data from the subscriber location, or for handling incoming calls and data in a corresponding manner, or may be arranged to handle traffic in both directions.

Since IP sessions, for transmitting IP packets, can be multiplexed simultaneously over a single subscriber line, providing the converter 230, and the telephone interface 240, enables the telephones in a household or small office to be used while a computer accesses the Internet using the computer interface and the multiplexer. However, under some circumstances, it is preferable not to convert the telephone call into IP packets, but instead to use the bridge 250. For example, it might be more cost effective, the voice quality may be better on one rather than the other, and one may be more reliable or connect more quickly than the other. One specific advantage is that it makes it easier to ensure continuous telephone service is available, even when a host PC is switched off, or when power is cut to converter or multiplex circuitry or hardware on which such functions are executed.

The functions shown in FIG. 2 need not be co-located. For example, the telephone interface and subscriber line interface could be located at the entry point where the subscriber telephone line enters the subscriberts location or site. The bridge could conveniently be located with the subscriber line interface and the telephone interface. Some of the multiplexer functions for handling IP packets from elsewhere could conveniently be implemented in software, running on a host computer, which may or may not be the same computer which sources the data being multiplexed with the telephone calls. As will be discussed below in more detail, it is possible to have the computer connected indirectly via the internal telephone line, rather than through its own direct connection to the multiplexer.

FIGS. 3, 4 alternative embodiments

In addition to all the features shown in FIG. 2, FIG. 3 has a telephone data interface, 260, which couples the internal telephone line to the subscriber telephone line via the multiplexer 200. This enables one or more computers connected to the internal telephone line to access on-line services through the multiplexer. This enables the computer directly connected to the multiplexer to continue to access on-line services concurrently.

In FIG. 4, the telephone data interface 260 is coupled to the multiplexer, but in such a way as to enable computers connected to the internal telephone line to pass data to the computer (the first computer), which is directly coupled to the computer interface 210. The telephone data interface could be coupled to both the computer interface and the subscriber line interface (not shown). This combination of the arrangements of FIGS. 3 and 4 would allow computers on the internal line to achieve the advantages of both FIGS. 3 and 4. The telephone interface and the telephone data interface may be arranged either so that only one computer or telephone terminal may be connected at one time, or techniques such as frequency division multiplexing may be used to enable the internal line to be used concurrently by a telephone device and a computer. The internal telephone line may be used as a local area network to enable multiple computers to communicate with each other.

The computers on the internal telephone line may use IP to communicate with on-line services, in which case the multiplexer might comprise a proxy server for relaying HTTP requests and responses in the form of HTML documents for example. The proxy server could reside on the first computer, connected directly to the multiplexer.

FIG. 5 alternative embodiment

In this embodiment, there is no direct connection to a computer, all computers are connected over the internal telephone line. Accordingly, there is a telephone data interface 260, but no computer interface 210. The telephone line interface and telephone data interface are arranged to be able to communicate simultaneously over the internal telephone line. Thus the multiplexer 200 can be used to multiplex a telephone call from the internal telephone line received via the telephone interface 240, together with a data transmission from one of the computers connected to the internal telephone line, received over the telephone data interface 260. A bridge corresponding to that shown in FIGS. 2 to 4 is optional, but is preferred for the reasons given above. One or more computers can be connected to the internal line, and thus access on line services while allowing telephony simultaneously over the internal line. Some of the advantages of this embodiment will be discussed below with reference to FIG. 14.

FIG. 6, HCTA Configuration Overview

FIG. 6 presents an overview of the configuration of the HTCA.

In this embodiment, the HCTA requires the subscriber (end-user) to have the following equipment:

A Personal Computer (PC) 380 to act as the HCTA controller. This PC, hereafter called the main PC, hosts the HCTA software and the HCTA hub card 340. Here, it is assumed that the HCTA hub is implemented as a card resident on the main PC. However, other implementations are also possible, as described below. This PC must also be equipped with a modem 330. The main PC is not dedicated to HCTA. HCTA activities should only consume a small fraction of the processing capacity of the CPU of the main PC. The modem is an example of part of the functions encompassed by the subscriber line interface. The HCTA software embodies the multiplexer function and demultiplexer function

mentioned above, and the monitoring function for determining if an IP session is established.

An HCTA hub card **340**. This card resides in the main PC. It implements a telephony interface, a voice processing DSP for Internet telephony, data communication control and the bridging of the in-house wiring to the PSTN. The hub should conform to the applicable telephony standards (regarding power, ringing, and so on). The hub card embodies the above mentioned converter, the bridge, and the telephone interface. It also embodies the computer interface, the telephone data interface, and the monitors for monitoring the subscriber line, and the internal line.

HCTA LAN adapter **360**. Each PC other than the main PC requires one such adapter to communicate with other PCs in the household and with the Internet.

An HCTA connection plaque **320**. This plaque is installed inside the house, near the drop, where the PTSN line enters in the household. The purpose of this plaque is to facilitate the re-connection of the in-house wiring for HCTA.

PSTN/ISP service requirements

In addition, the HCTA requires the end-user to subscribe to the following services:

From the local Telco a POTS line provisioned with a call forward busy feature. These services can be obtained by the end user himself, or by the Internet Call Waiting service provider as part of his service offer.

From the Internet Service Provider (ISP) **300**, an Internet service.

Local VoIP and ICW (Internet Call waiting) services **310**. These services can be offered by the end-user ISP, by the HCTA service provider, or by a third party ISP. These services should be available on a gateway deployed in the local calling area of the subscriber. The VoIP service provides IP/PSTN connectivity for telephony for the end user. VoIP services should preferably conform to the well known H323 standard. The ICW provides notification and disposition or diverting services for incoming calls to the subscriber when his line is busy.

The above are minimal requirements. The HCTA will function properly, and will achieve superior performance if the user possesses two lines, or can access the Internet at a higher speed than 56 kbps. The HCTA hub **340** and the modem **330** reside at the host PC, **380**. The HCTA hub is connected is to the PSTN/ISDN 80 by two wires (C). The HCTA is connected to all the devices hooked onto the inside wiring by two wires (A). These wires also connect together all the devices hooked onto the inside wiring. The HCTA is also connected to the modem by two wires. The wires (A) and (C) coexist in the existing 4-wire cables. This configuration requires no re-wiring and no additional wiring. However, it does require a modification of the connections at the drop. The role of the HCTA plaque, **320** located near the drop, is to facilitate the re-arrangement of the connections. If necessary, this re-arrangement at the drop can be avoided by modifying the RJ11 jacks or by inserting a small electrical connector between the RJ11 jacks and the devices hooked onto the inside wiring. More details are set out below.

The main PC is assumed to be always power-on. It is also assumed to be sufficiently powerful to host the HCTA control software. In event of power failure or of malfunction of the HCTA, the HTCA hub connects the in-house wiring (A) to the PSTN (C) and to the modem, thereby maintaining basic telephony, dial up access, and the in-house LAN. Secondary PCs **370** in the household, interface to the in-house wiring through an HTCA adapter **360**. This is a small box residing behind the PC, or a card in the PC, and will be described in more detail below. Telephones **350** are plugged into the internal line without any adapter.

FIG. **7** Network services operation for HCTA

FIG. **7** shows some features corresponding to those of FIG. **6**. In addition, a Central Office (CO) **400** is shown, where the subscriber line is linked to the rest of the PSTN. Other COs, **410** are shown to indicate that the ISP, VoIP and ICW servers need not be linked directly to the same CO as the subscriber, but may be linked to the PSTN at other COs. Assume the PC is connected to the Internet, and a call is arriving from the PSTN to the household. When the PC established its Internet connection, it notified the ICW server of its presence, and communicated to it its IP address.

When the call arrives at the CO **400**, it finds the subscriber line busy (as the PC is connected to the ISP). This activates the call forward busy feature provisioned on the subscriber line (as part of the ICW service). This feature causes the CO **400** to redirect the call to the ICW server, as indicated by the arrows in the PSTN region.

The ICW server receives the call. The signaling information informs it of the calling number and of the number where the call forward busy feature was triggered (the subscriber line). This allows the ICW server to identify the subscriber to which the call should be delivered.

The ICW server requests the VoIP server to establish a VoIP call with the IP address of the subscriber1s PC, and relays the incoming call to the VoIP server. The call can then be carried between the VoIP server and the subscriber1s PC over the IP session of this PC to the Internet. This part is similar to the operation of the above mentioned Phone Doubler product. Now suppose that the PC is connected to the Internet, and somebody in the house picks a phone to place a call. The HCTA hub card detects this event through its connection to the inside wiring (refer to FIG. **6**). It provides call supervision, and collects the digits dialed by the caller. It then requests the PC to establish a connection with the VoIP server over the Internet, and provides the VoIP gateway with the PSTN number that the caller wishes to reach. The VoIP gateway establishes the call to this number over the PSTN, and bridges it to the incoming link from the subscriber1s PC.

HCTA Operation Overview

Table 1 summarizes the principal actions of an embodiment of the HCTA hub to various likely events listed at the left of the table. Each column of the table gives the reactions to each event for a given state of the internal and subscriber lines. It is assumed there is only one subscriber line. The table is valid for embodiments with or without a direct computer interface. It is not necessarily valid for embodiments in which the LAN cannot be used simultaneously with telephony on the internal line.

TABLE 1

HCTA reactions to events

Event:	State: Fully idle	Non-IP data call	Internet session	Intercom	PSTN voice	Internet & intcm	Internet & voice
Incoming PSTN voice	Bridge it	Not seen	Ring then connect voip	Alert intercom users	Not seen	Alert intercom user	Alert phone user
Incoming VoIP	Not seen	Not seen	Pass to pc or ring phones	Not seen	Not seen	Alert intercom user	Alert phone user
Outgoing voice	VoIP or bridge	Busy: no action	Start voip	Busy: no action	Busy: no action	Busy: no action	Busy: no action
Outgoing Internet over lan	Set up call to ISP	Busy: no action	Mux sessions	Set up call to ISP	Alert phone users	Mux sessions	Mux sessions
Intercom	Ring phones	Ring phones	Ring phones	Busy: no action	Busy: no action	Busy: no action	Busy: no action
Lan	Route Data	Route Data	Route Data	Route Data	Route Data	Route Data	Route Data

Considering the first listed event, when an incoming PSTN call is sent to the subscriber, it is bridged to the internal line if possible, (column 1). If there is a non IP data call, or any other state which occupies either the internal or the subscriber line, the call will not be passed through but will be held or diverted. In the case of the existing state being a non-IP data call, or a PSTN voice call, the CO (Central Office) will divert the incoming call, and the HCTA hub will not see it. Otherwise, if the existing state is an intercom call (column 5), the hub can be aware of the incoming call and can try to alert the intercom user of the incoming call either by a voice announcement or tone. This may enable the subscriber to free the internal and subscriber lines to enable the incoming call to be accepted. If the existing state is one in which an Internet session is active (columns 4,7 and 8), the line is busy and the call will be diverted to the ICW server. In these states the hub will see an incoming VoIP call (processed as in row 2 of the table).

Regarding the second listed event, an incoming VoIP call, this will normally only reach the hub if an Internet session is active, unless the ISP has some special arrangement to call the subscriber when something is sent over the Internet to the IP address of the subscriber. In either case the hub can offer the call to any computer in the household which has VoIP capabilities. Alternatively, or as well, the hub can convert the call to a conventional call, and pass it to the internal line if it is free, or alert phone users by an announcement or tone.

Regarding the third row of the table, the start of an outgoing voice call, the hub will only see this event if the state is such that the internal line is not busy. If so, and if there is an Internet session occupying the subscriber line, there is a choice of converting the call to VoIP to enable it to be multiplexed with the existing Internet session, or alerting the computer user to request (or even force) that he end his Internet session, to free the subscriber line for the outgoing call to be passed out without conversion to IP. If the subscriber line is idle, the hub can choose whether to send the outgoing call as a VoIP call or a PSTN call. The choice may be governed by various factors, as discussed elsewhere in this document.

Considering the fourth row of the table, the event is an Internet access from a computer connected to the hub via the LAN. If the subscriber line is busy, and no Internet session is active, all the hub can do is try to free the line, e.g. by making an announcement to a phone user. If the subscriber line is free, the hub can set up a call to the ISP, to start the Internet session. If a session is already active, the new session can be multiplexed with the existing one.

The fifth row is the event of an intercom session starting. This is only possible if the internal line is free. If not, the new phone user may join in the existing call, without action from the hub. If the internal line is free, the hub needs to cause the internal phones to ring, when an intercom call is requested. The user may indicate a request for intercom by dialing a predetermined sequence of keys.

As shown by the sixth row of the table, the event of transmitting data over the LAN can occur regardless of the state of the lines.

As can be seen, the operation of the HCTA for voice communication depends on the state of three parameters:

Whether the inside wiring is currently supporting a telephone conversation.

Whether the line to the PSTN is busy or idle.

Whether the Main PC is connected to the Internet or not.

The relevant combinations of these states are summarized in Table 2. The operation of the product in each of these states is described in following six subsections.

Table 2

States of Telephony Parameters

State	Inside Wiring	PSTN line	Internet Access
1 n Fully idle	Idle	Idle	Not Connected
2 n Non-Internet data dial up	Idle	Busy	Not Connected
3 n Internet	Idle	Busy	Connected
4 n In-house telephony	Busy	Idle	Not Connected
5 n PSTN call active	Busy	Busy	Not Connected
6 n In-house telephony and Internet	Busy	Busy	Connected

1: Fully Idle State

When the inside wiring (A) and the PSTN line (C) are idle, the HCTA hub is continuously monitoring them to detect the occurrence of activity. The following events may occur:

a) A call arrives from the PSTN. The HCTA hub will answer the call, identify the called party, and then applies the ringing pattern associated with the called party on the in-house wiring. If the called party answers, the HCTA bridges the inside wiring to the PSTN line. Otherwise, the HCTA may offer to the caller the chance to leave a message. Depending on the identification of the caller, the HCTA may also immediately forward him to an announcement or to voice messaging.

For example, the HCTA may greet an incoming call by `ıYou have reached the Regnier Family, to whom do you wish to speak?î If the caller says ıto Jean pleaseî, or any sentence with the key word ıjeanî, Jeanıs distinct ringing pattern would be applied to the inside wiring. If the caller says ıto the master of the houseî, the HCTA would forward the call directly to voice messaging. Speech recognition enhances the HCTA by simplifying the caller interface.

However, is not mandatory to deliver the key benefits of HCTA. If speech recognition is too expensive or complex, a simpler menu driven interface can replace it.

b) Somebody in the house picks a phone to make a call: The HCTA will provide dial tone and will monitor the incoming digits. If the dialed digits indicate a request to the intercom service, the HCTA will monitor the incoming speech to detect the calling party requested. Upon hang up, the HCTA will apply the calling partyıs distinctive ringing pattern on the inside wiring. For example, mom could take a phone, press #, say `ıSophieî and hang up. Then Sophieıs distinctive ringing would be applied on the inside wiring. When Sophie answers, the ringing would stop, mom would pick up her phone, and they would be in communication. It may not be necessary for the caller to hang up, if ringing can be applied on the in-house wiring while the caller is listening.

c) If the dialed digits indicate a toll call, and if the HCTA has been set-up to establish toll calls by default over the Internet, the HCTA will launch an Internet session. Upon reception of all the dialed digits, the HCTA software on the main PC will establish a VoIP session with a remote gateway in the vicinity of the called party number.

d) If the dialed digits indicate a local call for which no special action needs to be taken, the HCTA will redial the digits onto the PSTN line, and will bridge the inside wiring to the PSTN line.

2: Non-Internet Data Dial-up State

This state occurs when the modem has established a data communication over the PSTN with a site from which an IP session from the Internet cannot be established with the subscriber. In this context, the Intercom service remains available as in the preceding section. However outgoing calls cannot be made, at least for the basic configuration with one POTS line. The restriction does not apply if the user has multiple lines, or an ISDN line, or an xDSL service.

Also, as the line is busy, the call forward busy feature on the end-user line will deflect incoming calls to the ICW service. There, the ICW service should play an announcement or apply a busy tone. Alternatively, the ICW service could offer a voice messaging service on call busy. Note that the ICW server knows that the call cannot be delivered to the end-user because the end-user would not be registered as active.

Occurrence of non-Internet data dial up should be rare, otherwise HCTA does not deliver its value, since the subscriber line cannot be used for dual purposes in this case.

The case of non Internet dial up from the internal line has not been discussed, as it can often be avoided by providing an HCTA adaptor for the device, to enable it to use the LAN to reach the HCTA hub.

3: Internet Session State

This state occurs when nobody in the house is using a telephone, and one or several PCs are hooked onto the Internet. The modem at the main PC has established a call with the ISP, tying the line.

Note that PCs users could be making calls directly from their PCs in this state. This would not generate telephony usage on the inside wiring. The following events may occur:

a) A call is arriving from the PSTN: The call will encounter a busy when it hits the end-user line, and will be deflected to the ICW server and then to the VoIP gateway. The VoIP gateway will establish a session with the main PC in the end-user premises, will transcode the call into VoIP format, and will deliver the call to the end-user PC. For this purpose, the end-user PC needs to register its IP address with the ICW server when it becomes online. The ICW server provides this address to the VoIP server to enable the VoIP call to be established to the end-user.

The PC can refuse to take the call, for instance if its bandwidth is too scarce to handle it. Then, the call would be sent to the ICW treatment upon busy, similarly as described for state **2**. Once at the PC and transformed into PSTN format, the PC answers the call, as described for state **1**.

b) Somebody in the house picks a phone to make a call: The HCTA will provide dial tone and will monitor the incoming digits. If the dialed digits indicate a request to the intercom service, the HCTA will behave as described for state **1**. If the dialed digits indicate a toll call, and if the HCTA has been set-up to establish toll calls by default over the Internet, the HCTA will establish a VoIP session with a remote gateway in the vicinity of the called party number, as described for state **1**. If the dialed digits indicate a local call, or indicate a toll call and the HCTA has been set-up to establish toll calls over the PSTN, the HCTA will establish a VoIP session with a local VoIP gateway. The local gateway will then relay the call to the PSTN.

4: In-house Telephony State:

In this state a conversation is being held between two phones in the household. The line to the PSTN is idle. The HCTA can answer an incoming call from the PSTN, but it cannot relay it on to the in-house wiring. The incoming call can be relayed to PCs, to be answered directly from one of the PCs, or it can be directed to announcement or to voice messaging.

The HCTA can also apply tones on the inside wiring to alert the household occupants currently on the phone to drop their conversation. When they free the in-house wiring, the incoming call from the PSTN can be relayed onto it. The alert tones can be called-party dependent, similarly as for the ringing patterns.

5: PSTN Call Active State:

A call is ongoing between a phone in the household and the PSTN. No PC can connect, and no other call can be established. Incoming calls are redirected to the ICW service by the call forward busy, where their treatment is similar as described for state **2**. The HCTA may be arranged to alert the phone user that another user is wanting to use the line, either for a voice call, or an Internet session for example. The phone user could be prompted to reconnect using VoIP, and the HCTA could do some or all of this reconnection automatically.

6: In-house Telephony and Internet Session State

One or more PCs are connected to the Internet, and either:

Two phones in the household are supporting an internal conversation (intercom), or;

One phone is involved in a call, and the call is relayed outside the household by the HCTA.

In this state, no outgoing call can be made from any phone. Outgoing calls can only be made from PCs, up to the 56 kbps bandwidth capacity. In this state, incoming calls would come through the VoIP gateway. These calls should be treated similarly as in state **4**. Namely, they could be answered by the HCTA, and relayed to PCs, to announcement or to voice messaging. The HCTA could also apply tones on the in-house wiring to alert the household occupant currently on the phone of the incoming call. If the call currently tying the inside wiring is terminated, the incoming call could then be relayed to the in-house wiring.

In-house LAN:

In normal mode, the main PC and the HCTA are power-on. They then offer an in-house LAN service and an Internet access service. The HCTA hub implements an Ethernet 10 baseT-like hub capability. This capability is supported by the existing in-house wiring, and coexists transparently with telephony signals (e.g., ringing, signaling, speech) that may also exist on the wires. Secondary PCs need an adapter to be on the in-house LAN. This adapter filters out telephony signals, and delivers the data stream to a standard PC port. All PCs connected to the LAN should be able to view the other PCs in their ınetwork neighborhoodi. They should also be able to view devices connected to the LAN. All PCs connected to the LAN should be able to view devices connected to other PCs, and be able to use them remotely. For instance, a PC should be able to download files to a printer attached to another PC.

Internet Access: The HCTA hub and main PC implement a multiplexing capability for Internet access. When the main PC connects to the Internet and establishes an IP session, other PCs can use this session as their channel to the Internet. This service is like a Proxy service. To the ISP, the other PCs appear simply as additional browser instances. When the main PC is connected to the Internet, secondary PCs access the Internet by connecting to the main PC over the in-house LAN. If a secondary PC wishes to access the Internet when the main PC is not connected, the secondary PC should be able to remotely instruct the main PC to connect to the Internet. Once the main PC is connected, the secondary PC accesses the Internet over the in-house LAN.

If the main PC uses the modem for a data connection other than to the ISP, Internet access is not available to the whole household.

Other Features

In several scenarios, incoming calls to the household are deflected to the ICW server. This may arise as a result of the call disposition selected by the user, or because it is the only option available (e.g., the line is busy). If the ICW server provides a messaging service on busy, the HCTA should check regularly for messages. If a message is waiting, the HCTA should apply a special tone on the telephones.

The HCTA should be capable of monitoring its integrity, and should be capable of raising alarms when it causes unacceptable degradation in telephony. In such events, the HCTA should stop all its telephony functions. Until restored, it should only connect the PSTN wiring (C) to the inside wiring (A) and to the modem.

In event of loss of power, the HCTA should connect PSTN wiring (C) to the inside wiring (A) and to the modem.

The HCTA software should run under Windows 95 and Windows NT.

The HCTA Hub

FIG. **8** presents an overview of the main components of the HCTA when implemented as a PC card. In this FIGURE, the ports are defined as follows:

HH: Household wiring.

CO1: Line to the central office.

CO2: Second line to the central office.

M1: Data connection to modem.

M2: Supplementary data connection to modem.

The elements of the HCTA depicted in FIG. **8** can be characterized as follows:

Bridge **500**: under the control of the HCTA software, the bridge establishes connections between the 2-wire ports: HH, CO1, CO2, **M1 M2** and Tel I/F. It may be implemented as a mechanical or solid state relay which will directly connect the internal line to the subscriber line when the host PC is switched off, or is in a fault state. Alternatively, the bridge could provide an indirect connection, for example through analog to digital converters and filtering software, or an opto electrical link to enable electrical isolation of the internal wiring.

Ethernet Hub **510**: implements a 10baseT-like Ethernet hub capability over existing in-house wiring. This Ethernet hub requires 2 wires to operate and coexists transparently with telephony signals on these wires. Given these constraints, the bandwidth may be lower than that of a true 10baseT Ethernet LAN. The Ethernet hub is always connected to the in-house wiring. This Ethernet hub is similar to the HomeRun HR1300HEC product by Tut Systems, and the reader is referred to descriptions of this product for more details. This product offers a 1.3 Mbps 10-baseT-like LAN, and meets all the above constraints.

Telephony Interface **530** (Tel I/F): provides full PSTN emulation via an analog telephone interface (dial tone, DTMF detection, call progress tones, ring voltage etc.). When connected to the in-house wiring by the bridge, the Tel I/F can drive all the phones in the house. The telephony interface also communicates with the HCTA software on the main PC to report events that it detects (e.g., off hook), or to implement instructions (e.g., dial a given number). The capabilities of the Tel I/F are well exemplified by the Internet Phone Jack product, and the reader is referred to their product description for more details.

VoIP **520**: the VoIP module may be implemented using a DSP for the transcoding of speech between PSTN and IP formats. This DSP is based on H323 codecs, and consists mainly of speech compression and echo cancellation. The H323 stack itself does not reside on the VoIP module, but resides on the main PC. The capabilities of the VoIP module are similar to the above mentioned Internet Phone Jack product, and the reader is referred to descriptions of this product for more details.

FIG. **9**, Hub operation with one line in ıMonitoringi mode.

In its basic configuration, the HCTA hub requires one POTS line to be connected to port CO1 and the inside wiring to be connected to port HH. All connections are 2-wires connections. It also requires a modem to be connected to port **M1**. Obviously, this modem is also connected to the main PC. Ports CO2 and **M2** are not used in this configuration. The HCTA hub configuration for operation with **1** POTS line is presented in FIG. **9**. This Figure depicts the HCTA in monitoring mode, when the household is fully idle (i.e., no call is under way and no PC is accessing the Internet). In this mode the bridge connects the HH to the Tel

I/F. The bridge also monitors itself the port CO1. Alternatively, the monitoring of CO1 can be done by connecting the modem to it, and by using the modem to detect the occurrence of activity, and to inform the HCTA control software in the main PC of this activity.

If somebody in the house picks a phone to make a call, the Tel I/F will detect this event. Depending on the dialed code and on the set-up of the HCTA operation, the following events may occur:

a) Intercom service invoked: for instance the user actions are: press # (to invoke the intercom service), speak `ıSophieî and hang up. The Tel I/F would recognize the # code and relay it to the HCTA software, informing it of the request for the intercom service. The Tel I/F would next relay the speech ıSophieî to the HCTA software for recognition. Upon recognition, the HCTA software would instruct the Tel I/F to apply Sophieıs distinctive ringing pattern on HH. When Sophie answers from some phone in the house, the caller would pick-up his phone and the communication would be established. It may not be necessary for the caller to hang up, if ringing can be applied while he listens.

b) Local call requested; for instance the user dials a local number. The Tel I/F will detect the incoming digits and will relay the digits to the HCTA control software. The HCTA will then instruct the bridge to connect HH to CO1. Depending on the speed at which the local call can be identified and the bridge reconfigured, the Tel I/F may have to regenerate the dialed number to the PSTN.

c) Toll call requested: for instance the user dials 1-xxx-xxxx. If the HCTA is configured to handle toll calls over the PSTN by default, the operations are as in the `ıLocal call requestedî event.

If the HCTA is configured to handle toll calls over the Internet by default, the Tel I/F will collect all digits and pass them to the HCTA control software. As soon as it learns that a toll call has been requested, the HCTA control software will instruct the bridge to connect CO1 and M1, and will instruct the modem to call the ISP. This will allow the ISP connection to be established while the user is completing his dialing, reducing his waiting time. When the dialed number is fully dialed and the ISP connection is established, the HCTA will establish a H323 session with a VoIP gateway in the vicinity of the dialed number, and will relay the dialed number to this VoIP gateway. The call will then be established over the Internet to the VoIP gateway, and over the PSTN from the VoIP gateway to the called number.

d) Incoming call arriving: the ringing signal on the port CO1 will be detected by the bridge, and reported to the HCTA control software. In return, the HCTA control software will instruct the bridge to connect CO1 to Tel I/F. The incoming call will then be answered by the HCTA to identify the called party. Upon identification, the bridge will connect HH to Tel I/F and Tel I/F will apply the called party distinctive ringing on HH. When the called party answers, the bridge will connect HH to CO1. This may require two 2-wire connections in Tel I/F to allow the incoming call to hear the ringing.

FIG. 10 HCTA Hub operating with one line in ıISP connection activeî mode.

FIG. 10 depicts the configuration of the HCTA hub when an Internet session is active. The bridge has established a path between CO1 and M1, and this path supports a call to the ISP. Depending on the dialed code and on the set-up of the HCTA operation, the following events may occur:

a) Intercom service invoked: the operations of the HCTA hub are similar to that in the ımonitoringî mode. Additionally, it could also notify the PCs of the call.

b) Outgoing call requested; for instance, somebody in the house picks a phone to make a call. The operation of the HCTA hub is similar to that for ıtoll call requestedî in monitoring mode, except that and ISP connection does not have to be established. Depending on the toll/local nature of the call and of the HCTA defaults, the call would be made to a local or a foreign VoIP gateway.

c) Incoming call arriving: as the line is busy, tied up by the ISP connection, the call must be arriving as a VoIP call over the ISP connection. The call is converted into the PSTN format by the VoIP module on the HCTA hub, and then answered by the PC, in a similar manner as described above for ıincoming call arrivingî in monitoring mode.

FIG. 11 HCTA Hub operating with one line in `ıPSTN telephony activeî mode

FIG. 11 depicts the configuration of the HCTA hub when a call is active between one of the household phones and the PSTN. A path is established between HH and CO1. In this configuration, it is not possible for PCs to access online services. PCs can only communicate locally over the LAN. As somebody in the house is already on the phone, the intercom service is also unavailable.

To allow PCs to access online services, the ongoing call needs to be terminated. Then, the main PC can establish an Internet connection. If necessary, the call that was terminated can then be re-established over the Internet, as the HCTA would now be in ıISP Connection activeî mode (see FIG. 10).

FIG. 12 HCTA Hub operating with two lines in `ıMonitoringî mode.

FIG. 12 depicts the HCTA hub in the configuration where the household is equipped with 2 lines. It is assumed that one of the lines is serving the modem attached to the main PC. When one or more PCs are online, incoming calls need not be delivered over VoIP. They can be delivered directly from the PSTN by connecting CO1 to HH. This avoids VoIP usage (and, potentially, charges) for local calls.

When one of the phones in the household is active in a call to the PSTN:

a) The call does not have to be dropped to allow PCs to access online services. The ISP access is set up through the second line hooked to the modem.

b) An incoming call (from CO1) can be deflected to CO2. If CO2 is idle, the main PC can handle it (e.g., to take a message, to present it to all PCs in the household, or to apply a tone on the inside wiring to alert the household of the incoming call). If CO2 is connected to the ISP, the call can also be delivered to the main PC through the VoIP service, and the main PC can handle it similarly.

FIG. 13 HCTA Hub operating with a xDSL modem in ıMonitoringî mode.

FIG. 13 depicts the HCTA hub in the configuration where the household is equipped with a high-speed xDSL modem **540**. It is assumed that the xDSL modem provides two distinct communication services: a conventional telephony service (such as POTS), and a high-speed data service. In this configuration, the xDSL high-speed data port connects directly into the main PC (for instance through an Ethernet card). The xDSL voice port connects to CO1. The differences in the operation of the HCTA hub with an xDSL modem to that of the basic one-line configuration are similar to those with the configuration with two lines. When one or

more PC are online, incoming calls need not be delivered over VoIP. They can be delivered directly from the PSTN by connecting CO1 to HH.

When one of the phones in the household is active in a call to the PSTN, the call does not have to be dropped to allow PCs to access online services. Furthermore, an incoming call can be redirected to the main PC, if online, to handle it (e.g., to take a message or to present it to all PCs in the household). In addition, due to the high bandwidth available with xDSL and over the LAN, this configuration enables many calls to be made concurrently. However, all but one of these calls must be made from PCs. Another possible arrangement is with two lines and an inverse multiplexer, (to enable a single data stream to be split over the two lines) or with an ISDN line.

HCTA adapter

The HCTA adapter is a 10BaseT-like Internet Network Interface Card. The HomeRun HR1300ISA adapter card produced by Tut Systems is an example which can be used. It enables LAN signals to be frequency division multiplexed over the internal telephone line at frequencies high enough that they will not interfere with telephony signals.

HCTA software functions

The main PC hosts the HCTA software responsible for the following functions:

a) Driver for the DSP card: this software module controls the installation and configuration of the HCTA hub card. It should also be able to control the upgrade of the HCTA hub card, for instance, to download new DSP software loads as standards evolve and codecs are improved.

b) Protocol host for H323 and TCP/IP: this software module provides H323 protocol services for VoIP connection set-up and management. Similarly, it provides TCP/IP protocol services for IP session set-up and management. These protocol services are expected to reuse largely existing software (e.g., Microsoft Netmeeting).

c) Multiplexing of IP sessions: this software module provides the capability to multiplex several IP sessions over a common ISP connection. These IP sessions can be made locally from the main PC, or can be made remotely over the in-house LAN from secondary PCs. This service is similar to that provided by the Internet Lanbridge product mentioned above. This software also implements a bandwidth protection mechanism for VoIP sessions.

d) Execution of telephony service logic: this software module is responsible for the control of the telephony operations of the HCTA hub card. Upon notification of events from the card (e.g., incoming call from PSTN, digits dialed in the in-house wiring, etc), it executes the HCTA telephony service logic (for intercom, Internet toll calls, etc.).

e) Control of the PC card bridge configuration: as dictated by the HCTA service logic, this software module provides the HCTA card with the appropriate telephony and bridge configuration controls.

f) Auxiliary services: these regroup the services that are not at the core of the operation of the HCTA, but that are used to support its core services. For instance, speech recognition is an auxiliary service that is used to support the core incoming call, intercom and outgoing calling services.

g) Installation and service management shields: this software module is responsible to support the end-user in installing and managing the HCTA.

h) The software on the main PC may also include a PC-based telephony interface, for instance for screen-based notification of incoming calls.

Software on other PCs

Further or secondary PCs connected to the LAN using an adapter described above, have HCTA LAN software enabling the end-user to install and manage their LAN connection to the Ethernet hub on the main PC. The software on the secondary PCs may also include a PC-based telephony interface, for instance for screen-based notification of incoming calls.

The HCTA Plaque for reconnecting the internal line to the subscriber line

The HCTA plaque is a small piece of hardware screwed or nailed in the wall inside the house near the telephone drop. Its role is to facilitate the re-arrangement of the connections for HCTA. It may not be necessary if as in older houses, the point where the subscriber line meets the various branches of the internal line, is accessible in one of the rooms of the house. In this case, the PC which hosts the HCTA adapter can be connected directly in between the internal line and the subscriber line, without needing to use the four wires of one of the branches of the internal wiring to enable this PC to be inserted between the internal line and the subscriber line. The plaque may not be needed for embodiments in which the HCTA hub is implemented as a stand alone peripheral, which can be located away from its host PC, or for embodiments corresponding to FIG. **5**, in which all PCs are connected over the internal line. The plaque has three ports, one for the subscriber line (CO port), one for the internal line (Inside wiring port), and one (PC port) for the branch of the internal wiring, ıhost PC wiringi, leading to the PC hosting the hub. Within the plaque, two of the wires from the PC port are connected to the CO port. The other two wires from the PC port are connected to the Inside wiring port. To install the plaque, the inside wiring needs to be disconnected from the subscriber line, and the host PC wiring needs to be separated from the other branches of the internal line.

FIG. **14**, Configuration overview of standalone embodiment

The implementation of the HCTA described above with a host PC which has a direct computer interface to access the subscriber line without using the internal line has some advantages. It can minimize the cost of the product and it reuses devices that the subscriber may already possess, such as a modem. However, an alternative implementation of HCTA involves a hub in the form of a standalone box based on the arrangement shown in FIG. **5** with no PC having a direct interface. All PCs would be connected over the internal telephone wiring. One or more of the PCs could still retain the host function of being able to configure the standalone box remotely. The standalone box could be located near the drop, provided a source of power is available. Such an implementation would differ from that presented in this document in the following ways:

a) The HCTA hub would integrate all the run-time capabilities of HCTA. It would also integrate an on-board modem, or an interface to a separate local modem.

b) Configuration of the HCTA hub would be done from the main PC, but the main PC would not be involved in its operation. This would free the main PC from the requirement of being always on.

c) The HCTA hub would require a CPU, memory and an I/O port for autonomous operation. This will add to its cost. On the other hand, the HCTA hub may not require a DSP chip, as the VoIP processing could be done by the on-board CPU chip.

d) Installation would be simple. It would consist in screwing the HCTA to the wall near the drop, and in connecting the CO line to one port, and the in-house wiring to another. However, it would require availability of a source of power near the drop.

e) The main PC would need to be equipped with a network interface card for communication over the in-house LAN.

f) It would be easier to install HCTA for 2-line operation or enable a later upgrade for such operation, assuming both subscriber lines are available at the drop. This is because there is now no need for one or more separate branches of the internal wiring for the host PC Such branches may otherwise require some rewiring, to separate them from the rest of the internal wiring.

This alternative implementation has simpler installation requirements. It requires the user to screw the HCTA to the wall near the drop, and to plug it to a source of power. It is more expensive in terms of the HCTA product itself, but it may save the visit of a professional installer, hence may be cheaper overall to the end subscriber. To implement the hub in standalone form, hardware corresponding to that of the PC card would be needed, as well as an onboard dedicated host processor to run the HCTA software functions described above.

Other Variations

Although the embodiments described have used wireline for the subscriber line and the internal line, the apparatus and method is in principle applicable to a wireless subscriber service, e.g. a fixed access or mobile cellular service, and to an internal line which uses wireless techniques. Hence the terms subscriber telephone line and internal line are intended to encompass these alternatives. Other variations within the scope of the claims will be apparent to persons of average skill in the art, and are not intended to be excluded.

What is claimed is:

1. Apparatus for use at a subscriber site with a subscriber telephone line connecting the subscriber site and a telephone network, for simultaneously sending a telephone call from a PSTN compatible telephone terminal on the subscriber site, and IP packets from a first computer, over the subscriber telephone line, the apparatus comprising:

a converter having a PSTN compatible telephone line interface for coupling to the PSTN compatible telephone terminal, the converter being for converting signals from the PSTN compatible telephone terminal into IP packets representing the telephone call; and

a multiplexer coupled to the converter, and having a computer interface for coupling to the first computer, and a subscriber line interface for coupling to the subscriber telephone line, the multiplexer being arranged for sending simultaneously the IP packets representing the telephone call and those from the computer, along the subscriber telephone line;

the apparatus being arranged to handle telephone calls without conversion to IP packets, between the telephone terminal and the subscriber telephone line when the subscriber telephone line is not used for carrying IP packets.

2. The apparatus of claim 1, further comprising a bridge for selectively coupling the subscriber telephone line to an internal telephone line to which the telephone terminal is attached.

3. The apparatus of claim 1, the telephone terminal being coupled to the apparatus via an internal telephone line, the apparatus further comprising a telephone data interface coupled to the multiplexer and arranged for coupling to the internal telephone line, for receiving data from a further computer coupled to the internal telephone line, and distinguishing this data from analog signals from the telephone terminals, the multiplexer being arranged to send the data from the further computer along the subscriber telephone line.

4. The apparatus of claim 1, the telephone terminal being coupled to the apparatus via an internal telephone line, the apparatus further comprising a telephone data interface coupled to the multiplexer and for coupling to the computer interface, for receiving data from a further computer coupled to the internal telephone line, the multiplexer being arranged to send the data from the further computer to the first computer.

5. The apparatus of claim 4, the telephone line interface and the telephone data interface being arranged to communicate over the internal telephone line simultaneously.

6. The apparatus of claim 1, the telephone line interface further comprising an internal line monitor for determining the state of telephony activity from the telephone terminal, the apparatus being arranged to control a telephone call in response to the determination.

7. The apparatus of claim 1 further comprising a subscriber line monitor for determining the state of the subscriber telephone line the apparatus being arranged to control a telephone call in response to the determination.

8. The apparatus of claim 1, further comprising an IP session monitor for determining when there is an IP session already existing, the apparatus being arranged to control a telephone call in response to this determination.

9. The apparatus of claim 8, further comprising an internal line monitor for determining the state of telephony activity from the telephone terminal, and a subscriber line monitor for determining the state of the subscriber telephone line, the apparatus being arranged to control the call additionally on the basis of the outputs of the internal line monitor and the subscriber line monitor.

10. The apparatus of claim 1, the telephone line interface being arranged to determine what number has been dialed, and establish the call as a VoIP call, or establish the call as a conventional telephone call, on the basis of the number dialed.

11. The apparatus of claim 1, further comprising an internal line monitor, for determining the state of telephony activity from the telephone terminal, the apparatus being arranged to send a busy indication in response to an incoming call for the telephone terminal, if an IP session is established and if the internal line monitor determines that the telephone terminal is busy.

12. The apparatus of claim 1 being arranged to prompt a caller of an incoming call to indicate a party that he intends to reach, and the telephone interface being arranged to process the call on the basis of the party indicated by the caller.

13. The apparatus of claim 12, the telephone line interface being arranged to output an alert specific to the party indicated by the caller.

14. The apparatus of claim 1, the multiplexer further comprising a second subscriber line interface for coupling to a second subscriber line.

15. The apparatus of claim 1, the multiplexer further comprising a second subscriber line interface for handling a second connection over the subscriber telephone line, for use with a high bandwidth coupler for multiplexing the second connection over the subscriber telephone line.

16. Apparatus for use at a subscriber site with a subscriber telephone line connecting the subscriber site and a telephone network, for simultaneously receiving a telephone call to a PSTN compatible telephone terminal on the subscriber site, and IP packets directed to a first computer, both sent over the subscriber telephone line, the apparatus comprising:

a converter having a PSTN compatible telephone line interface for coupling to the PSTN compatible tele-

phone terminal, the converter being for converting from IP packets representing the telephone call into signals for the PSTN compatible telephone terminal; and

a demultiplexer coupled to the converter, and having a computer interface for coupling to the first computer, and a subscriber line interface for coupling to the subscriber telephone line, the demultiplexer being arranged for receiving simultaneously the IP packets representing the telephone call and those for the computer, over the subscriber telephone line;

the apparatus being arranged to handle telephone calls without conversion to IP packets, between the telephone terminal and the subscriber telephone line when the subscriber telephone line is not used for carrying IP packets.

17. A method of simultaneously sending a telephone call from a PSTN compatible telephone terminal on a subscriber site and IP packets from a first computer, over a subscriber telephone line from the subscriber site, the method comprising the steps of:

converting signals from the PSTN compatible telephone terminal into IP packets representing the telephone call;

using a multiplexer coupled to the converter, and having a computer interface for coupling to the first computer, and a subscriber line interface for coupling to the subscriber telephone line, to send simultaneously the IP packets representing the telephone call and those from the computer, along the subscriber telephone line; and

handling telephone calls without conversion to IP packets, between the telephone terminal and the subscriber telephone line when the subscriber telephone line is not used for carrying IP packets.

18. Software stored on a computer readable medium for carrying out a method of simultaneously sending a telephone call from a PSTN compatible telephone terminal on a subscriber site and IP packets from a first computer, over a subscriber telephone line from the subscriber site, the method comprising the steps of:

converting signals from the PSTN compatible telephone terminal into IP packets representing the telephone call;

using a multiplexer coupled to the converter, and having a computer interface for coupling to the first computer, and a subscriber line interface for coupling to the subscriber telephone line, to send simultaneously the IP packets representing the telephone call and those from the computer, along the subscriber telephone line; and

handling telephone calls without conversion to IP packets, between the telephone terminal and the subscriber telephone line when the subscriber telephone line is not used for carrying IP packets.

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US-PAT-NO: 5267305

DOCUMENT-IDENTIFIER: US 5267305 A
See image for Certificate of Correction

TITLE: Transparent inband signaling

---------- KWIC ---------

Abstract Text - ABTX (1):
A method and apparatus for transparent inband signaling during a telephone conversation provides an advantageous way of enabling a party to a telephone conversation to signal a telecommunications system to perform desired functions. A person signaling a system with a general inband signal, such as DTMF, can do so transparently, that is without the other party hearing the inband signals used. This innovation is based on detecting the inband signal, muting the inband signal to prevent the other party from hearing it, and determining which party generated it while allowing the conversation to continue substantially uninterrupted. The inband signal is delayed prior to muting for a short period of time in an alternate embodiment in order to compensate for the time delay in reliably detecting DTMF signals thus mute the inband signal more completely.

TITLE - TI (1):
Transparent inband signaling

Brief Summary Text - BSTX (13):
The invention may be defined in various terms. According to one definition of the present invention, it is a method for the detection and processing of

inband telecommunications signals, such that inband signals generated by one person in a conversation are rendered transparent, i.e. unnoticeable to the other person, and to process the signals to initiate the desired telecommunications system functions. This method in combination with a security feature also prevents a party to a conversation not authorized to do so from initiating telecommunications system functions.

Brief Summary Text - BSTX (15):

A telecommunications system with transparent inband signaling can intercept inband signals such as DTMF tones generated by one person during a telephone conversation in progress. Those tones can be intercepted and blocked so as to make them inaudible to the other person in the conversation. Blasting the other party in the ear with DTMF tones is not only very disruptive and irritating, but can also cause the other person to wonder if the system has malfunctioned. Transparent inband signaling operates unobtrusively and does not disturb conversations, which is an important advantage of the invention.

Brief Summary Text - BSTX (16):

In some applications, remote users or telecommuters may have to enter a password or authorization code to gain remote access to a telecommunications system. The source discrimination inherent in transparent inband signaling determines which party to a conversation has generated an inband signal. If an authorized remote agent generates the appropriate DTMF code, then the system can initiate the desired functions, such as, call the supervisor, etc.

Drawing Description Text - DRTX (68):

FIG. 66 is an overall general functional block diagram of elements needed to implement transparent inband signaling within a trunk interface of a telecommunications system;

Drawing Description Text - DRTX (69):
FIG. 67 is an overall general functional block diagram of elements needed for an implementation of transparent inband signaling within a trunk interface of a telecommunications system and incorporating analog and digital circuitry;

Drawing Description Text - DRTX (70):
FIG. 68 is an overall general functional block diagram of elements needed to implement transparent inband signaling within a trunk interface of a telecommunications system with a matrix switch;

Drawing Description Text - DRTX (71):
FIG. 69 is an overall general functional block diagram of elements needed to implement transparent inband signaling within a trunk interface of a telecommunications system, incorporating a 2-wire to 4-wire converter;

Drawing Description Text - DRTX (72):
FIG. 70 is an overall general functional block diagram of elements needed to implement transparent inband signaling within a trunk interface of a digital telecommunications system;

Drawing Description Text - DRTX (73):
FIG. 71 is an overall general functional block diagram of elements for an implementation of transparent inband signaling within a trunk interface of a telecommunications system with a 2-wire to 4-wire converter, incorporating a

delay line in the inbound portion of the signal path;

Drawing Description Text - DRTX (74):
FIG. 72 is an overall general functional block diagram of elements needed
for an implementation of transparent inband signaling within a trunk interface
of a telecommunications system, incorporating a delay line within a hybrid
circuit;

Drawing Description Text - DRTX (76):
FIG. 74 is an overall general functional block diagram of elements needed
for an alternate implementation of transparent inband signaling within a trunk
interface of a telecommunications system, incorporating a delay line within a
hybrid circuit;

Drawing Description Text - DRTX (77):
FIG. 75 is an overall general functional block diagram of elements needed to
implement transparent inband signaling within a trunk interface of a
telecommunications systems, incorporating analog and digital circuitry and a
delay line within a hybrid circuit;

Drawing Description Text - DRTX (78):
FIG. 76 is an overall general functional block diagram of a signal path
through a telecommunications system with interfaces with transparent inband
signaling capability.

Drawing Description Text - DRTX (79):
FIG. 77 is an overall general functional block diagram of elements needed
for an alternate implementation of transparent inband signaling within a trunk
interface of a telecommunications system and incorporating analog and digital

circuitry.

Detailed Description Text - DETX (230):
DETAILED DESCRIPTION OF TRANSPARENT INBAND SIGNALING

Detailed Description Text - DETX (231):
FIG. 64 shows in general block diagram form, those elements of system 1A shown in FIG. 1 involved in DTMF signal processing, to provide transparent inband signaling capability. It should be understood that in this and subsequent figures showing the elements required to implement transparent inband signaling, the circuit elements needed for trunk control such as ring detector, loop current detector, etc. and associated trunk interface control logic have been omitted since the design and use of such elements is well known to those skilled in the art.

Detailed Description Text - DETX (233):
The operation of this invention as embodied in FIG 64 is shown in FIG. 65 which is a more detailed flow chart of certain operations generally referred to in block 47G of FIG. 47A, in particular operations for processing DTMF signals to provide transparent inband signaling capability.

Detailed Description Text - DETX (239):
When an agent hits a DTMF key on his telephone, while in a conversation with a caller, the caller will not notice the DTMF, since he will not hear more than the first 20 milliseconds of a detected DTMF tone, and the rest of the tone will be muted. The sequence of events shown in FIG. 65 in checking for DTMF signals during a conversation in progress between an agent and a caller make possible the remote activation of system features from a standard telephone

with a DTMF pad. An agent while conversing with the caller, can perform
various system functions, such as placing the caller on hold, calling a
supervisor, etc. Pressing the "1" key places the caller on hold. Pressing the
"2" key will signal the supervisor and put the caller on hold. In the event of
an emergency, the agent can press "5" to signal the supervisor to monitor the
conversation in progress. The agent's ability to activate system functions
using the DTMF signals during a conversation without the caller hearing the
signaling is an important feature and advantage of transparent inband
signaling.

Detailed Description Text - DETX (241):
The method and apparatus for transparent inband signaling shown in FIGS. 64
and 65 can be incorporated in various types of telecommunications systems using
trunk or line interfaces. Telecommunications trunks are generally used to
connect a system to other systems or to the public switched network.
Telecommunication lines are used to connect a system, such as a switch, to
other systems or to end equipment, such as telephone instruments. A
communications pathway through a system requires an internal path or connection
to be made between a trunk or line interface and another trunk or line
interface. The internal connections between interfaces or ports of a system
can be hardwired or through a switching arrangement. Telecommunications
systems such as PBXs and ACDs use internal matrix switches, such as crosspoint
or digital matrix switches.

Detailed Description Text - DETX (242):
FIG. 66 is an arrangement of those elements needed for implementing

transparent inband signaling in a general purpose trunk interface of a telecommunications system. Transparent inband signaling capability can be incorporated in the telecommunications interfaces of various types of systems such as PBXs, ACDs, Voice Response Units and other types of systems. A remote user on the public switched network using DTMF tones to remotely activate functions could be connected to a telecommunications system through such a trunk interface. The sequence of events in the operation of the apparatus in FIG. 66 is the same as the sequence of events shown in FIG. 65. Upon the detection of DTMF signals, DTMF Detector 20C produces an Early Steering (ESt) signal (CTL 167) as part of the DTMF bus, which is used by Controlling Logic and Decoders 64A to generate CTL 44 Audio Connect Control. Audio Connect Control CTL 44 is selectively used to control Switch 14C for DTMF source discrimination and DTMF muting. The trunk interface shown in FIG. 66 is typically connected to other subsystems such as a trunk interface, line interface, a matrix switch or other subsystem through internal Connection 66A in order to establish a communications pathway through a telecommunications system. Just as the trunk interface shown in FIG. 65 incorporates transparent inband signaling, a similar apparatus for transparent inband signaling could be incorporated in a line interface.

Detailed Description Text - DETX (245):

FIG. 69 demonstrates an embodiment to implement the transparent inband signaling of the present invention in a trunk interface incorporating a 2-wire to 4-wire converter. Converter 69A separates the incoming and outgoing signals propagating in opposite directions in the 2-wire telecommunications signal

path. Upon detection of a DTMF tone, an ESt signal (CTL 167) is sent by Detector 20C to CPU 68C via Bus Interface 68B. CPU 68C then issues a command to Matrix Switch 69B to open the appropriate ports of the Matrix for source discrimination and muting.

Detailed Description Text - DETX (246):

FIG. 70 shows a block diagram of elements for an embodiment to implement transparent inband signaling of the present invention within a digital telecommunications system. In such a system, the two wire analog signals are converted to four wire digital signals by Coder-Decoder (Codec) 67A. Codec 67A separates the incoming and outgoing signals propagating in opposite directions on the 2-wire telecommunications signal path. Digital Signal Processor (DSP) 70A is controlled and programmed by Central Processing Unit (CPU) 68C to detect inband signals. The control and programming of DSPs to detect DTMF signals is well known and reference is made to the following application notes: "Add DTMF Generations and Decoding to DSP-Microprocessor Designs" by P. Mock, Volume I, 1986, pp. 543-547 and "General-Purpose Tone Decoding and DTMF Detection," by C Marven, Vol. 2, 1990, pp. 423-526, from "Digital Signal Processing Applications with the TMS 320" published by Texas Instruments Corp. Once a DTMF signal is detected during a conversation, a DTMF Sense signal is sent via the Output Bus of DSP 70A to CPU 68C which generates a control signal to Matrix Switch 67B to open the signal path for source discrimination and to block propagation of the detected inband DTMF signals. The output bus of DSP 70A also indicates to CPU 68C which DTMF tones have been detected for the initiation of desired system functions.

Detailed Description Text - DETX (247):
FIG. 71 is a block diagram of elements for an alternate implementation of transparent inband signaling within a telecommunications system, incorporating a delay line in the inbound portion of the signal path. As in the implementation shown in FIG. 69, a 2-wire to 4-wire converter is used to separate the incoming and outgoing signals. DTMF detectors require a finite amount of time in order to reliably detect DTMF signals. False triggering of the detectors is possible because of the audio frequencies found in human speech. Such false triggering of DTMF detectors is known as "talk-off." Some DTMF detectors require 10 millisecond of time for the earliest detection of DTMF. Some detectors require 30 milliseconds before a DTMF data valid signal is generated. In a situation where it takes 20 milliseconds to know if you have valid DTMF being received, then there is at least a 20 millisecond delay in opening the signal path either for source discrimination or for the blocking of DTMF tones. This means that the first 20 milliseconds of a DTMF signal will not be blocked and will be heard by the other party in the conversation. Twenty milliseconds of DTMF does not really sound like a DTMF tone but sounds more like a short noise burst.

Detailed Description Text - DETX (248):
FIG. 71 shows an implementation of transparent inband signaling where the time lag between the first incidence of DTMF and the first opening of the signal path have been eliminated, by introducing a time delay using Delay 71A in the transmit side of the signal path of the trunk interface. Delay line 71A can be implemented using well known and readily available analog or digital

delay technology, such as charge coupled devices (CCDs) forming a bucket brigade delay or a digital memory delay unit.

Detailed Description Text - DETX (250):

An alternate approach in the use of a delay line to implement transparent inband signaling can be seen in FIG. 72. In this 2-wire interface, DTMF detector 20C checks for DTMF tones and sends out an Early Steering (ESt) CTL 167 signal on the DTMF Bus within 20 milliseconds of the start of detection of a DTMF tone. The ESt signal is sent to Control Logic and Decoders 64A which outputs an Audio Connect Control (CTL 44) signal to Switch 14C to open the signal path to block the detected DTMF signals, as needed for source discrimination and muting. The bidirectional signal passing through the interface is separated into the inbound and outbound signals within Hybrid with Delay 72A, whose configuration can be seen in FIG. 73.

Detailed Description Text - DETX (251):

The Hybrid Circuit shown in FIG. 73 separates the two signals present in the 2-wire interface of FIG. 72. The two signals are the two sides of the conversation passing through the interface. The general arrangement of active and passive elements shown in FIG. 73 as a hybrid circuit is well known to those skilled in the art. Delay line 73I is introduced into the circuit for the purpose of an alternate implementation of transparent inband signaling. The Hybrid with Delay shown in FIG. 73 provides for a delay, such as 20 milliseconds, in Delay 73I for the inbound portion of the conversation. Op amp 73G, delay 73I and op amp 73J provides a unidirectional path for the incoming signal propagating from left to right. Op amps 73B and 73A provide a

unidirectional path from right to left for the outgoing signal. Balance filters 73C and 73F and Attennuators 73D and 73E are used to internally balance the hybrid for stable operation by adjusting the phase and frequency response and level of the inverted signals so as to achieve maximum echo cancellation. The inverted incoming signal is available at the output of op amp 73G at the junction labeled 73H and the inverted outgoing signal is available at the output of op amp 73B. Op amp 73G derives the incoming signal by summing the inverted outgoing signal provided by op amp 73B with the bidirectional signal also sent to 73G. Op amp 73B derives the outgoing signal by summing the inverted incoming signal with the bidirectional signal also fed to 73B. The time delay introduced by delay 73I in the inbound signal is substantially equal to the time that it takes the DTMF Detector 20C to reliably detect DTMF. Thus, Switch 14C can be opened in response to the generated Audio connect control CTL 44 signal at the same time as the DTMF tone arrives, blocking its further propagation.

Detailed Description Text - DETX (252):

FIG. 74 shows an alternate implementation of transparent inband signaling using a delay line within a hybrid. In FIG. 74 the telecommunications signal is split by the Hybrid with Delay 72A using the circuit shown in FIG. 73. DTMF detector 20C senses the incoming signal at point 73H prior to delay 73I. In this configuration, the DTMF detector is detecting inbound DTMF. The time delay introduced by Delay 73I in the inbound signal is about equal to the time it takes to reliably detect DTMF. Thus, Switch 14C can be opened at the same time as the DTMF tone arrives. Switch 14C can be opened as needed for source

discrimination and for the blocking of DTMF.

Detailed Description Text - DETX (254):
FIG. 76 shows a communications path through a system with interfaces having transparent inband signaling implemented using inband signal detection on the inbound portion of the communications signal at each interface.

Detailed Description Text - DETX (255):
The interfaces shown in FIGS. 67, 69, 70, 71, and 75 all utilize methods whereby the detection of DTMF takes place on the incoming portion of the two-way signal which is on the interface. If such an interface with transparent inband signaling is used in both Interfaces 76B and 76D in FIG. 76, then no incoming DTMF will reach Interconnection 76C from either direction. In implementing Transparent Inband Signaling in the interfaces shown in FIG. 76, it will be sufficient to open the incoming signal path in both interfaces, as needed, to block any incoming DTMF, since no detectable DTMF will arrive at the DTMF Detector or DSP on the outbound portion of the signal in each interface.

Detailed Description Text - DETX (257):
FIG. 77 is similar to FIG. 67, except that only the Transmit (TX) output of Codec 67A is opened to block the propagation of incoming DTMF signals. Similar changes could be made to FIGS. 69, 70, 71 and 75 to block only incoming DTMF in configurations where interfaces with transparent inband signaling are used as in FIG. 76.

Detailed Description Text - DETX (258):
An interface where the incoming and outgoing signals are separated, such as

the one shown in FIG. 77 can also function properly in a system where the other interface in the communications path does not have transparent inband signaling capability, if in the conversion from two-wire to four-wire, sufficient separation between the incoming and outgoing signals can be achieved.

Detailed Description Text - DETX (259):

Transparent inband signaling provides an inaudible method for signaling a telecommunications device or system during a conversation in progress in order to initiate desired system functions.

Claims Text - CLTX (1):

1. An apparatus for transparent inband signaling coincident with an active telephone conversation conducted by parties with at least one party having a user station capable of generating inband signals, the apparatus comprising:

Claims Text - CLTX (45):

28. A method for transparent inband signaling coincident with an active telephone conversation conducted by parties with at least one party having a user station capable of generating inband signals, the method comprising the steps of:

Claims Text - CLTX (55):

34. An apparatus for transparent inband signaling coincident with an active telephone conversation conducted by parties with at least one party having a user station capable of generating inband signals, the apparatus comprising:

US-PAT-NO: 5937040

DOCUMENT-IDENTIFIER: US 5937040 A

TITLE: Method and apparatus for using a D-channel for displaying user data

---------- KWIC ---------

Detailed Description Text - DETX (4):

In FIG. 1, the voice information is transmitted to the display phone 12 via one or more B-channels 16 of an ISDN interface 18. However, the use of ISDN telecommunications is not critical to the invention. Other links which include one or more user data channels in addition to a signaling channel may be substituted for the ISDN interface 18. The signaling channel in FIG. 1 is a D-channel 20 that is utilized in the conventional manner to transmit connectivity-related messages, such as CONNECT, ACKNOWLEDGE, DISCONNECT, and RELEASE signals. In addition to the exchange of connectivity-related signals that are transparent to the caller, the D-channel is conventionally used to transmit connectivity-related signals that are apparent to the caller by presentation at a display 22 of the telephone 12. Such signals are DISPLAY IE messages or a container (i.e., envelope) having user-to-user information, and may be used to identify a party calling the display phone. In further addition to the signaling information that is conventionally exchanged via a signaling channel, such as the D-channel, the present invention employs the signaling channel to transmit user data that is typically reserved for user data channels, such as the B-channel 16. In the embodiment of

FIG. 1, the user data
information that is transmitted over the D-channel is
character strings
indicative of the menu options of the IVR unit 10.

US-PAT-NO: 6345047

DOCUMENT-IDENTIFIER: US 6345047 B1

TITLE: Computer telephony adapter and method

---------- KWIC ---------

Detailed Description Text - DETX (90):

In normal mode, the main PC and the HCTA are power-on. They then offer an in-house LAN service and an Internet access service. The HCTA hub implements an Ethernet 10 baseT-like hub capability. This capability is supported by the existing in-house wiring, and coexists transparently with telephony signals (e.g., ringing, signaling, speech) that may also exist on the wires. Secondary PCs need an adapter to be on the in-house LAN. This adapter filters out telephony signals, and delivers the data stream to a standard PC port. All PCs connected to the LAN should be able to view the other PCs in their 1network neighborhood1. They should also be able to view devices connected to the LAN.
All PCs connected to the LAN should be able to view devices connected to other PCs, and be able to use them remotely. For instance, a PC should be able to download files to a printer attached to another PC.

Detailed Description Text - DETX (107):

Ethernet Hub 510: implements a 10baseT-like Ethernet hub capability over existing in-house wiring. This Ethernet hub requires 2 wires to operate and coexists transparently with telephony signals on these wires. Given these constraints, the bandwidth may be lower than that of a true 10baseT Ethernet LAN. The Ethernet hub is always connected to the in-house

wiring. This
Ethernet hub is similar to the HomeRun HR1300HEC product by
Tut Systems, and
the reader is referred to descriptions of this product for
more details. This
product offers a 1.3 Mbps 10-baseT-like LAN, and meets all
the above
constraints.

Detailed Description Text - DETX (143):
f) Auxiliary services: these regroup the services that are
not at the core
of the operation of the HCTA, but that are used to support
its core services.
For instance, speech recognition is an auxiliary service that
is used to
support the core incoming call, intercom and outgoing calling
services.